TABLE OF CONTENTS

SECTION **PAGE**

LIST OF FIGURES

LIST OF TABLES

SECTION 6
UNIQUE CAPABILITIES AND REQUIREMENTS

Section 6 contains requirements that are unique to Deployed (Tactical) systems in Section 6.1 and Classified systems in Section 6.2. The unique requirements are modifications to, or additions to, the overall requirements defined in Section 5, Unified Capabilities Product Requirements.

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6.1 UNIQUE DEPLOYED (TACTICAL) REQUIREMENTS

6.1.1 Introduction

This section defines unique Deployed (Tactical) requirements. Section 5.1, Requirements Categories and Language, provides the general definition of requirements terminology used. Deployed (Tactical) requirements that are common to Fixed requirements for DVX systems are set forth in UCR 2008, Section 5.2, Circuit-Switched Capabilities and Features.

This section was created by the Executive Agent for Theater Joint Tactical Networks (EA-TJTN) to identify the Tactical requirements in consonance with responsibilities assigned by ASD(NII). In addition, the U.S. Military Communications-Electronics Board (MCEB) tasked the Theater Joint Tactical Networks Configuration Control Board (TJTNCCB) to develop Tactical interoperability requirements as certification criteria for joint networked-communications systems. In pursuing acquisition initiatives, COCOMs, military services, and defense agencies shall use this section as a guideline for the purchase of COTS equipment, as well as for the development of systems that need to interface in deployed networks. The Tactical networked-communications community of the DoD shall adhere to this section to comply with DODI 8100.3, "DOD Voice Networks."

6.1.1.1 Purpose

This section defines the unique Deployed requirements that are not contained in UCR 2008, Section 5.2, Circuit-Switched Capabilities and Features, and the Fixed requirements that need to be modified to support Deployed users. This section consolidates interoperability certification requirements to the maximum extent possible and incorporates them as part of requirements for the overarching GIG in support of network-centric warfare. This section provides guidance for satisfying the certification requirements for Deployed voice systems used as part of an Operational Area Network (OAN), which is the deployed extension of the GIG. This section also defines other UCR elements applicable to the Deployed community, and serves as a ready reference to be used by the JITC when writing the Deployed annex to the Generic Switch Test Plan (GSTP).

6.1.1.2 Applicability

The requirements described in this section apply to NEs, LANs when used in Deployed (Tactical) environments, ~~and~~ Deployed Cellular Voice Exchange (DCVX) Systems, and LSCs.

6.1.1.3 Definitions

Definitions and Acronyms are provided in Appendix A, Definitions, Abbreviations and Acronyms, and References.

6.1.2 Circuit-Switched-Based Deployable Network Designs and Components

Circuit-Switched-based deployable requirements are defined in UCR 2008, Sections 6.1.2 and 6.1.3.

6.1.3 Deployed Voice Quality

The desired objective for Deployed voice quality is an MOS of 4.0 or greater, but it is realized that the network may operate under less than ideal conditions. The requirements provided in the following paragraphs are the minimally acceptable values under the conditions specified. The MOS calculation will assume the use of G.729 with 20 ms samples for the purpose of SLAs.

6.1.4 Deployed -NE General Requirements

~~This content has been moved to~~Deployed NE General Requirements are found in Section 5.9.

6.1.5 Deployed LANs

6.1.5.1 Overview

Tactical Operations Centers (TOCs) and other deployed enclaves operate under austere conditions, rely on a Deployed power supply/grid, and may be restrictive in the size, weight, and packing requirements. The Deployed LAN and the backbone and transmission components operate from the same Deployed power source. It is extremely difficult to approach the availability and power backup requirements mandated on the Fixed infrastructure with its commercial-grade power supply and r fixed operating environment.

The ASLAN requirements defined in Section 5.3.1, Assured Services Local Area Network Infrastructure, represent the optimal LAN design and Deployed users are encouraged to implement their requirements whenever possible. However, operational realities often preclude the deployment of highly redundant components and multiple backup power sources.

6.1.6 DCVX System Requirements

6.1.6.1 Introduction and Purpose

The following sections describe the requirements that shall be met by all deployed DCVX systems for them to be certified and used in the OAN tier of the Global GIG. Requirements are defined at the system level as well as for the various components that make up the cellular system, including protocol requirements. The DCVX is a cellular system with MUFs, and therefore, is not the same as commercially deployed Mobile Cellular Systems (MCS).

It is recognized that not all components may be needed for a specific application. The requirements discussed in this section are similar to those for the SMEO, DVX, or PBX1 (See UCR 2008 Section 5.2) and dependent on the network configuration and specific authorized gateway connection.

6.1.6.2 Applicability

The requirements within this section are applicable to the following:

- All DCVX systems that connect directly or indirectly to the DISN voice systems including the DSN, DRSN Secure Phone Gateways, and/or commercial PSTN.
- Procured or leased cellular systems that connect to any DISN service gateway. Commercial cellular services are not allowed to be connected to DISN service gateways.
- Procured or leased cellular systems using leased cellular frequencies that connect to any DISN service gateway.

The current version of the UCR is the governing requirements document that takes precedence over the explicit or implicit requirements of subsidiary or reference documents, standards, and specifications. In the event of a conflict, the explicit requirements of the UCR take precedence over the explicit or implicit requirements of any other requirements document except for those requirements specified in the documents listed in Section 6.1.6.3, Policy and Reference Documents.

6.1.6.3 Policy and Reference Documents

The following policy and instruction documents, in conjunction with the current version of the UCR, will be used as the basis for APL certification:

1. Policy for the use of commercial wireless devices, services, and technologies in the DoD GIG, as outlined in DODD 8100.2. This directive further promotes joint interoperability using open standards throughout DoD for commercial wireless services, devices, and technological implementations.

2. "Wireless Priority Service (WPS) Industry Requirements for the Full Operating Capability (FOC) for CDMA-Based Systems – Home Location Register (HLR)," Issue 1, 04 June 2004.

3. "Wireless Priority Service (WPS) Industry Requirements for the Full Operating Capability (FOC) for GSM-Based Systems," Issue 2, January 2004.

4. 3G TS 24.067 V3.0.0 (1999-05), 3rd Generation Partnership Project; Technical Specification Group core Network; enhanced MLPP (eMLPP) – Stage 3.

6.1.6.4 DCVX System Overview

The DCVX systems provide wireless mobile communication services with MUFs and draw their Strategic services by means of approved DoD authorized gateway switching systems only. The DCVX can be connected to a DVX-C or connected directly to the DSN via Deployed communications as described in CJCSM 6231.01b. The DCVX systems also may be interconnected with other cellular telephone systems, excluding commercial systems unless the commercial system is procured or leased for DoD usage and is operating in isolated mode from other commercial cellular systems. The DCVX also may be connected to both the DSN and the systems described previously simultaneously to E2E MUF, such as MLPP.

When placed in a joint Deployed environment, the DCVX will have the capability to connect to DSN and between other DCVXs and DVXs, via Tactical voice, using UCR-defined protocols such as ISDN PRI, MLPP PRI (T1.619a), and/or CCS7. The CCS7 over IP using IETF SIGTRAN may be used only in connecting DCVXs together on DoD IP networks within the Tactical OAN. However, only ISDN PRI may connect to the commercial PSTN and/or other non-Government networks. A DCVX system also may be configured to interconnect at the network transmission level with other DCVX systems to provide roaming capability outside the local home base cellular network for supported terminal devices.

Cellular terminal devices, often referred to as mobile subscribers cellular handsets, PDAs, "blackberries," and any other user cellular end item devices, commercial or Government developed, may connect to commercial cellular systems when operating outside the transmission range of the DCVX.

Additionally, the cellular terminal devices may have the capability to interface with other wireless networks (IEEE 802.11 and IEEE 802.16) and commercial cellular service when not supported by a DCVX. Actual employment of this additional cellular terminal device capability will be by command approval only, in the OAN.

6.1.6.4.1 DCVX Components

The DCVX is comprised of three major components with multiple subcomponents and/or devices:

- Terminal device(s)
  - Cellular or mobile handset
  - PDA
  - "Blackberry"
  - Government-developed terminal
  - Other commercial cellular devices

- Base Station Subsystem (BSS)
  - Base Transceiver Station (BTS)
  - Base Station Controller (BSC)
  - Cell tower(s) with radio transceiver(s)

- Deployed Mobile Switching Center (DMSC)
  - Mobile Switching Office (MSO)
  - Home Location Register (HLR)
  - Visitor Location Register (VLR)
  - Authentication Center (AUC)
  - Equipment Identity Register (EIR)

6.1.6.5 DCVX Operation

The DCVX functions and provides mobile cellular services similar to standard commercial cellular systems with the addition of MUFs. It is based on a two-way cellular radio system that interconnects cell phones with other cell phones and landline stations. When used, the DCVX will provide full mobile cellular coverage in designated deployed environments; this includes training, exercise, and operational missions within COCOM AORs or specific geographic areas. User voice, data, and related communications via terminal devices will be similar to landline wired DSN or commercial services. Except for the inherent characteristics of radio transmission, basic service features between the two systems will be similar and transparent to the users. After full mature architectural implementation, the DCVX will function as a wireless adjunct and extension of the joint OAN tier of the GIG. The following configurations, illustrated in Figure 6.1.6-1, Deployed Cellular Voice Exchange Generic Design, define the operational deployment options of a DCVX.

6.1.6.5.1 Subtended Deployment Connection

For a "subtended deployed connection," the DCVX will only interface with DSN voice services using an existing authorized gateway switch, i.e. DVX-C, to connect to the Tactical transport network, with the one or more of the following interfaces:

- ISDN PRI (T1/E1)
- MLPP ISDN PRI (T1/E1) **[Conditional]**

6.1.6.5.2 Direct DSN Deployment Connection

For a direct DSN connection, the DCVX will use the "direct connection" configuration to the Tactical transport network with one or more of the following interfaces:

- ISDN PRI (T1/E1)
- MLPP ISDN PRI (T1/E1) **[Conditional]**
- CCS7 [Conditional]
- IP AS-SIP (signaling and associated bearer channel) **[Conditional]**

The DCVX can directly connect to the DSN by either TDM or IP, but not both simultaneously.

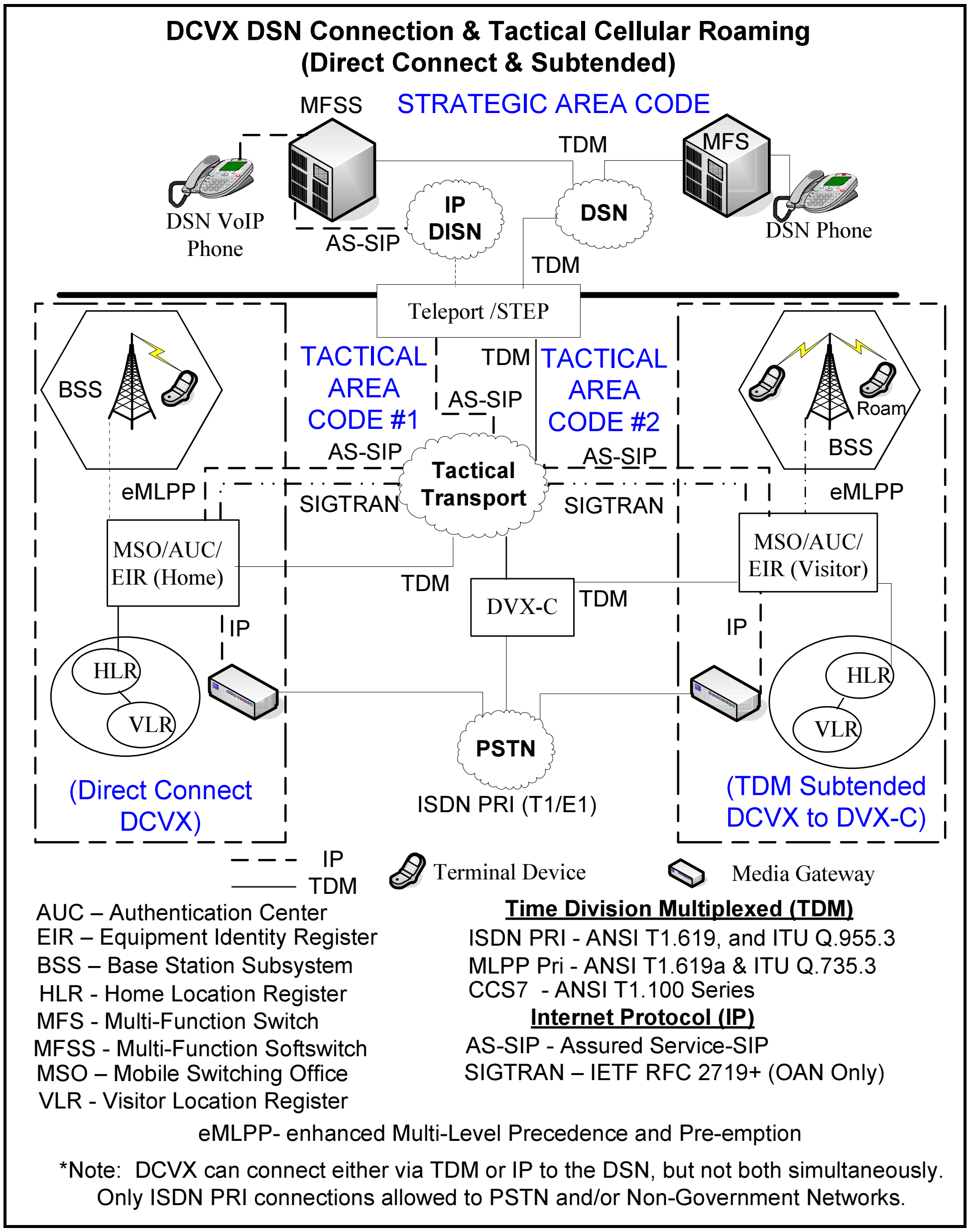


Figure 6.1.6-1. Deployed Cellular Voice Exchange Generic Design

6.1.6.5.3 Networked DCVX Deployment

When a DCVX is deployed in a networked DCVX configuration, a large deployed unit or multiple deployed units within the Tactical OAN may be connected with one or more HLR routing tables configured to support cellular terminal device roaming capabilities per the interconnections previously described.

For networked DCVXs within the Tactical OAN in support of terminal device roaming capability, the DCVX configuration to the Deployed transport network will be with one or more of the following interfaces:

- ISDN PRI (T1/E1)
- MLPP ISDN PRI (T1/E1) **[Conditional]**
- CCS7 [Conditional]
- IP AS-SIP (signaling and associated bearer channel) **[Conditional]**
- SIGTRAN (CCS7 over IP) **[Conditional]**

The extent of terminal device roaming capability will depend on the number and type of interconnections made between the DCVXs within the Tactical OAN and switch lookup routing table updates in the DCVXs themselves.

For all connection variations, the DVCX will connect to the PSTN and/or other non-Government networks via TDM ISDN PRI (T1/E1) only.

6.1.6.5.4 Standalone DCVX Deployment

When a DCVX is used in standalone configuration, the only area served is a deployed unit establishing a Joint Task Force (JTF) and its command, control, communications, and computers (C4) infrastructure. There is no DSN or PSTN access and no roaming beyond the deployed local network unit cell towers of its area of operation. The DCVX operates solely in an isolated mode.

6.1.6.6 General Description of Cellular Mobile Features and Technologies

6.1.6.6.1 Priority Access Service/Wireless Priority Service

Priority access service provides the logical means for authorized mobile users to queue to the front and obtain priority access to the next available channel in a wireless call path. The goal of the WPS is to provide an E2E OAN-wide wireless priority communications capability to key military personnel during natural or man-made disasters. The WPS is an enhancement to basic cellular service. The full WPS capability can provide priority handling from mobile call origination, through the network, and all the way to the call destination.

The WPS is invoked by keying a special access number (*272) before the destination number on cellular instruments that have been classmarked for the WPS feature. A WPS user may be assigned one of five priority levels (i.e., 1, 2, 3, 4, or 5), with “1” being the highest priority level and “5” being the lowest. Each priority level has user-qualifying criteria that may track that for MLPP in DSN.

When a WPS call is queued for a radio traffic channel, from a cellular user, and no channel is available, the call is queued according to (1) the highest Priority Access Service (PAS) priority first, and (2) queue entry time (i.e., earliest call first) within the same priority. If the queue for the call sector is full and the caller's priority is determined to be higher than the level of the lowest priority caller in the queue, then the most recent WPS entry shall be removed, with the new WPS call request queued in accordance with (1) and (2), above.

6.1.6.6.2 DoD Global System for Mobile Cellular Band

The dedicated DoD Global System for Mobile (GSM) band is from 1755 MHz to 1835 MHz, which is a subset of the commercial DCS-1800 band. The remaining Government-owned frequency ranges are 1755 MHz to 1785 MHz for the uplink and 1805 MHz to 1850 MHz for the downlink. There are no non-DoD regulatory challenges associated with the use of the GSM band. The band has been approved for exclusive DoD use and is not authorized for use by any other entity. This band will be used for both voice and data applications to support unique DoD requirements.

The band benefits are only effective in a CONUS environment; however, the DoD GSM may be used OCONUS with specific host country(s) authorization. The normal DoD frequency allocation process shall be followed to allow system operation within this band, and CC/S/A planners must ensure that an alternative solution is available before deployment as part of the planning process.

6.1.6.6.3 Precedence and Preemption

Precedence and preemption can only be implemented in a DoD network. This service has two parts: precedence and preemption. Precedence involves assigning a priority level to a call (wireless or wired). Preemption involves the seizing of a communications channel that is in use by a lower precedence level caller, in the absence of an idle channel. In the DCVX, the Precedence and Preemption capability is conditional. Precedence and preemption may be provided by enacting eMLPP or a vendor proprietary version that performs precedence and preemption in the DCVX between the terminal device and the cellular switch. The eMLPP is a cellular version of MLPP. In either version, precedence will be invoked by keying defined digits before dialing the destination number on cellular instruments that have been classmarked for this service. Precedence will function jointly in combination with WPS and will perform E2E as an adjunct to regular MLPP service on the wired DSN. However, in either of the provided versions, if available in the DCVX, eMLPP or vendor proprietary, the connection to the DSN will be MLPP PRI (T1.619a) and/or CCS7 or AS-SIP.

Mobile systems, as currently designed, provide a maximum of seven priority levels. The two highest levels (A and B) are reserved for network internal use, (e.g., for emergency calls or the network-related service configurations for specific voice broadcast or voice group call services).

The second highest level (B) can be used for network internal use or optionally, depending on regional requirements, for subscription. These two levels (A and B) can only be used locally, that is, in the domain of one DCVX. The other five priority levels are offered for subscription and can be applied globally (e.g., on interswitch trunks) if supported by all related NEs, and for interworking with ISDN networks providing the MLPP service. The seven eMLPP priority levels and their respective mapping to MLPP are defined as follows:

- **A** highest, for network internal use
- **B** for network internal use or, optionally, for subscription
- **0** for subscription: FLASH-OVERRIDE
- **1** for subscription: FLASH
- **2** for subscription: IMMEDIATE
- **3** for subscription: PRIORITY
- **4** lowest, for subscription: ROUTINE

Levels A and B shall be mapped to level "0" for priority treatment outside of the DCVX area in which they are applied. The vendor-proprietary version will support the five precedence levels as specified for DSN MLPP.

6.1.6.6.4 Code Division Multiple Access Mobile Systems

Mobile Code Division Multiple Access (CDMA) technology uses spread spectrum telecommunications techniques in which a signal is transmitted in a bandwidth considerably greater than the frequency content of the original information. The latest technology today is based on third generation (3G) that allows high and fast bandwidth, generically called Evolution-Data Optimized (EVDO or EV-DO). This capability supports data usage of the terminal device to allow data connections to DoD networks and future possible use of VoIP softphone on terminal devices when connected to commercial networks for extension of DSN single number presence.

6.1.6.6.5 GSM Communications Mobile Systems

Early technology for GSM allowed the use for Time Division Multiple Access (TDMA) technology. The TDMA allows several users to share the same frequency. It is the most popular standard for mobile phones in the world. The ubiquity of the GSM standard makes international roaming very common with "roaming agreements" between mobile phone operators. The latest GSM standard is based on an open standard that is developed by the Third Generation Partnership Project (3GPP).

6.1.6.6.6 Secure Communications Interoperability Protocol

The SCIP is the NSA-approved secure voice and data encryption protocol used by DoD, U.S. Government agencies, and civilian authorities. The SCIP is used by NATO and coalition partners also to provide secure voice interoperability between the United States and authorized foreign entities. Application of SCIP is described in detail within UCR 2008, Section 5.2.12.6, DoD Secure Communications Devices.

6.1.6.7 DCVX Requirements Terminology

Requirements terminology is defined in Section 5.1.4, General Requirement Language.

6.1.6.8 DCVX General Requirements

6.1.6.8.1 Coverage and Signaling Strength

[Required] The signal strength shall not be less than the current GSM and CDMA authorized international standards and specifications. The GSM and CDMA technology are spectrum based; therefore, GSM/CDMA band, coverage, signal strength, and power are the basis for planned "Area of Support." Environment, weather, geography, topography, and adjacent spectrums are elements that must be considered when applying the basis for Area of Support. For testing purposes, the generic set of parameters presented in Table 6.1.6-1, Current Cellular Systems Parameters, shall be used for JITC certification either by testing and/or as determined by JITC.

6.1.6.8.2 Protocol/Format

[Required] The DCVX shall support at least one or more of the following protocols:

- GSM/GPRS (2.5G, 3G, 3GSM, GSM Edge)
- WCDMA
- CDMA2000
- CDMA 1XRTT
- UMTS
- EVDO (or EV-DO)

Table 6.1.6-1. Current Cellular Systems Parameters

DCVX (DMSC, BSS) GSM/GPRS	
Bands	As provided by standards and/or DoD GSM Cellular Band (e.g., 450 MHz, 850MHz, 900MHz, and 1900 MHz)
Specification on Coverage	As provided by standards (e.g., ITU-R 2.5G, 3G, 3GSM, GSM Edge (www.itu.int/publications))
Distance Transmit/ Receive	Up to 25 miles depending on topology/man-made structures, and frequencies also determine coverage parameters.
DCVX (DMSC, BSS) CDMA	
Bands	As provided by standards (e.g., 450 MHz, 700 MHz, 800 MHz, 850 MHz, 900 MHz, 1700 MHz, 1800 MHz, 1900 MHz, and 2100 MHz)
Specification on coverage	As provided by standards (e.g., Telecommunication Industry Association, IS95, 3GPP2, IMT-2000, CDMA 1XRTT, CDMA2000) (www.tiaonline.org)
Distance Transmit/ Receive	Up to 32 miles depending on topology/man-made structures, and frequencies also determine coverage parameters.
TERMINAL DEVICE	
Bands	As provided by standards (CDMA/GSM) and/or DoD GSM Cellular (e.g., 450 MHz, 700 MHz, 800 MHz, 850 MHz, 900 MHz, 1700 MHz, 1800 MHz, 1900 MHz, and 2100 MHz)
CDMA Specification	As provided by standards (e.g., CDMA(IS95), CDMA2000, CDMA 1XRTT and CDMA 1xEVDO)
GSM Specification	As provided by standards (e.g., GSM (GSM 02.07 Tech. Spec.(ver.7.1.0 Rel. 1998)), 2.5G, 3G, 3GSM, GSM Edge)
Distance Transmit/ Receive	Up to 8 miles depending on topology/man-made structures, and frequencies also determine coverage parameters.

6.1.6.8.3 MOS and Measuring Methodology

[Required] The DCVX shall support the minimum MOS scores as defined in Section 5.3.3, Network Infrastructure End-to-End Performance Requirements, or better as measured in any 5-minute interval using P.862 testing standard. The baseline test environment shall be while operating in an open air, clear obstruction, line-of-site environment with the specific requirements as outlined in Table 6.1.6-1. Based on the results, the estimated MOS performance range will be extrapolated and provided in the vendor LOC based on the Base Station Antenna operating at or near full power mode and, at a minimum, operating height of 80 feet. The values provided in the vendor LOC will be included in the APL report. Refer to Section 6.1.6.14, Submission of Wireless Systems to UCCO for DSN Connection Request, concerning guidelines on submitting the cellular engineering analysis package.

6.1.6.8.4 Availability

[Required] The DCVX shall have an availability of 99.99 percent, which includes scheduled maintenance.

6.1.6.8.5 Encryption

[Required] The DCVX must provide appropriate radio and network transport bandwidth to support secure calls via SCIP, other NSA-accredited encryption scheme(s), and/or other required accredited encryption schemes as defined in the appropriate STIGs for cellular to support terminal device encryption requirements in Section 6.1.6.9.4, Terminal Device Encryption.

[Required] The DCVX shall support SCIP, other NSA-accredited encryption scheme(s), and/or required accredited encryption schemes as defined in the appropriate STIGs for cellular. The STIG-required encryption shall be provided to secure the wireless call, as a minimum, if SCIP and/or other NSA-accredited encryption schemes are not provided. The DCVX that supports SCIP (a.k.a. terminal device) will be required to go secure E2E with another SCIP Phone and/or via a SCIP Gateway if AS-SIP is used while the DCVX supports the establishment and maintaining of the secure call.

[Conditional] The DCVX may have the capability to provide secure SCIP gateway functions.

6.1.6.8.6 Calling Features

6.1.6.8.6.1 Call Waiting Feature Requirement

The CW feature interacts with MLPP. If a precedence and preemption capability is available in the DCVX, the MLPP interactions must meet the requirements described in Section 6.1.6.8.10.1, Precedence Call Waiting. Call Waiting is a feature where a line in the talking state is alerted by a CW tone when another call is attempting to complete to that line. A CW tone is only audible to the line with the CW feature activated.

[Required] The CW feature shall generate a CW tone only audible to the line with the CW feature activated.

[Required] The Cancel CW feature is required when CW is active. The user must be able to cancel the CW service. Cancel CW is a feature that allows the user with CW service to inhibit the operation of CW for one call. The user dials the Cancel CW code, obtains recall dial tone, and places a call normally. During this call, the CW service shall be inactive so that anyone calling the CW user shall receive the normal busy treatment, and no CW tones shall interrupt the user’s call.

6.1.6.8.6.2 Three-Way Calling Requirement

The TWC feature interacts with MLPP. If a precedence and preemption capability is provided in the DCVX, the MLPP interactions must meet the requirements described in Section 6.1.6.8.10.2, Precedence TWC.

[Conditional] The TWC is a feature that allows a station in the talking state to add a third party to the call without operator assistance. To add a third party to the call, the TWC customer places the other party on hold, receives recall dial tone, dials the third party's telephone number, and then takes the first line off hold to establish the TWC connection. This may occur any time after the completion of dialing the second number joining the TWC. After the TWC connection has been established, the customer with the service activated may disconnect the last party added. The customer with the service activated may terminate the TWC call by disconnecting. If either of the other two parties hangs up while the service-activating customer remains off-hook, the TWC is returned to a two-party connection between the remaining parties.

[Conditional] The terminal device may support signaling to allow TWC.

6.1.6.8.6.3 Conference Calling

The Conference Calling feature is conditional because it interacts with MLPP. If a precedence and preemption and conference calling capabilities are provided in the DCVX, the MLPP interactions must meet the requirements described in Section 6.1.6.8.10.3, Precedence Conference Calling.

[Conditional] This feature allows the user to establish a conference call involving up to six conferees (including the user). This feature is requested via an access code.

[Conditional] The terminal device may support signaling to allow conference calling.

6.1.6.8.7 Roaming

[Conditional] The DCVX system may only support roaming to one or more DCVXs within the Tactical OAN. Network connections with commercial cellular systems in support of roaming are not allowed. Roaming shall meet the Global Block Numbering Plan (GBNP) requirements, as specified for the DVX-C in Section 6.1.3.3**,** Deployed **(**Tactical) Routing and Numbering.

6.1.6.8.8 Precedence and Preemption

The DCVX may support preemption and precedence under the following conditions:

1. **[Conditional]** The DCVX may support the cellular version of MLPP, called eMLPP, and/or a proprietary methodology. When precedence and preemption are available, the TDM interface to the DSN network and/or the supporting DVX-C shall support MLPP PRI and/or CCS7 as described in Section 6.1.6.11.5.1, MSO MLPP Trunks and Interfaces.

2. **[Conditional]** The DCVX will support a preemption and precedence capability under one or more of the following conditions.

    a. The DCVX supports GSM in the DoD GSM cellular band as described in Section 6.1.6.6.2, DoD GSM Cellular Band.

    b. The DCVX supports the use of leased cellular frequency in one of the bands and protocol(s) listed in Table 6.1.6-1, Current Cellular Systems Parameters.

    c. The DCVX supports one or more of the cellular bands and protocol(s), as described in Table 6.1.6-1, Current Cellular Systems Parameters, in an OCONUS environment, where the local Forces-Status Agreement allows eMLPP/proprietary version operation.

    d. The DCVX supports one or more of the cellular bands and protocol(s), as described in Table 6.1.6-1, Current Cellular Systems Parameters, dependent on the operational environment and usage of cellular frequencies allowed by local and/or national civilian authorities.

6.1.6.8.9 Precedence Capability Terminal Device Activation/Deactivation

[Conditional] If a precedence and preemption capability is provided in the DCVX, the DCVX may be capable of providing any supported terminal device the user's Precedence Class Table Assigned features for providing said features to the terminal device based on the user entering a specified PIN number on same said terminal device. The DCVX will assign to the terminal device all the user's precedence capability as defined in the switches class features table(s). This will allow the user to make precedence calls from different terminal devices other than the one assigned or provided to the user. Additionally, the precedence features assigned to that active terminal device can be turned off by re-entering the same or different PIN number on the said terminal device. The precedence capability user's activation/deactivation PIN number may be stored in the DCVX or in another database accessible by the DCVX to validate the user's PIN number(s) associated with the user's precedence capability. The user's precedence activation or deactivation PIN number may be assigned and/or user settable after given an initial assigned PIN number.

6.1.6.8.10 Precedence and Preemption Calling Features

[Conditional] If a precedence and preemption capability is provided in the DCVX, then under the following calling features, once a higher precedence call has been connected to the terminal device and the higher precedence call is in progress, the calling party of lower precedence will receive a notification that the lower precedence call was rejected.

6.1.6.8.10.1 Precedence CW

[Conditional] The following Precedence CW treatments shall apply to precedence levels of priority and above if the Precedence and Preemption capability is provided in the DCVX.

6.1.6.8.10.1.1 Busy with Higher Precedence Call

[Required] If the precedence level of the incoming call is lower than the existing MLPP call, precedence CW shall be invoked. In an active call, if the incoming call is priority precedence or above, the precedence CW tone shall be applied to the called party.

6.1.6.8.10.1.2 Busy with Equal Precedence Call

[Required] The DCVX shall provide the precedence CW signal to the called station. The DCVX shall apply this signal regardless of other programmed features, such as call forwarding on busy or caller ID. The called station shall be able to place the current active call on hold, or disconnect the current active call and answer the incoming call.

6.1.6.8.10.1.3 Busy with Lower Precedence Call

[Required] The DCVX shall preempt the active call. The active busy station shall receive continuous preemption tone until an on-hook signal is received and the other party shall receive preemption tone for a minimum of three seconds. After the current call is terminated and the terminal device is idle, the station to which the precedence call is directed shall be provided precedence notification, described in UCR 2008, Section 5.2, Circuit-Switched Capabilities and Features, or comparable vibration cadence. The station shall be connected to the preempting call after going off-hook.

6.1.6.8.10.1.4 No Answer

[Required] If, after receiving the precedence CW signal, the busy called station does not answer the incoming DSN call within the maximum programmed time interval, the switch shall treat the call in accordance with UCR 2008, Section 5.2.2.3, Precedence Call Diversion.

6.1.6.8.10.2 Precedence TWC

[Conditional] If precedence and preemption and TWC are provided in the DCVX, the following TWC requirements apply:

1. **[Required]** In TWC, each call shall have its own precedence level. When a TWC is established, each connection shall maintain its assigned precedence level. Each connection of a call resulting from a split operation shall maintain the precedence level that it was assigned upon being added to the TWC.

2. **[Required]** The DCVX shall class mark the originator of the TWC at the highest precedence level of the two segments of the call. Incoming calls to lines participating in TWC that have a higher precedence than the higher of the two segments shall preempt unless the call is marked non-preempt able.

3. **[Required]** When a higher precedence call is placed to any one of the TWC participants, that participant receives the preemption tone. The other two parties shall receive a conference disconnect tone. This tone indicates to the other parties that one of the other TWC participants is being preempted.

4. **[Required]** In a TWC call where each connection is established at a different precedence level, the precedence level of the participant who initiated the TWC call shall be assigned the highest precedence of the two connections.

6.1.6.8.10.3 Precedence Conference Calling

[Conditional] If precedence and preemption and conference calling are provided in the DCVX, then the following precedence conference calling requirement is required:

1. **[Required]** All addresses shall be processed at a precedence level equal to that precedence level dialed by the conference originator.

   a. If all conference bridges are busy, ROUTINE precedence conference call attempts shall be connected to "Line Busy" tone, and call attempts at precedence levels above the ROUTINE precedence shall re-examine all conference bridges on a preemptive basis.

   b. A conference bridge that is busy at the lowest level of precedence stored for all units shall be preempted for a higher precedence conference call.

c. When a conference bridge is preempted, a 2-second burst of preemption tone shall be provided to the conferees on the existing conference. The existing connections to the bridge shall be dropped, and the bridge shall send an on-hook signal automatically to the associated switch ports to permit the new connections to be established.

d. Where the requesting precedence level is equal to or lower than, the existing conference, the connection shall be denied and the caller shall be provided a BPA.

6.1.6.8.10.4 Voice Mail

The Voice Mail feature interacts with MLPP. If precedence and preemption capability and voice mail are provided in the DCVX or voice mail added externally, the MLPP interactions must meet the requirements described in UCR 2008, Section 5.2.2.3, Precedence Call Diversion.

[Conditional] The DCVX may provide ROUTINE calls only voice mail capability for users. Additional features such as message forwarding and others may be provided in addition to basic voice mail capability provided they do not interfere with precedence and preemption if capability is provided in the switch.

6.1.6.8.10.4.1 Precedence and Preemption Interaction with Voice Mail

[Conditional] If precedence and preemption is provided in the DCVX and voice mail capability is provided internally to the DCVX or connected externally to the DCVX as an adjunct, the following requirement applies:

[Required] The DCVX shall divert all precedence calls above ROUTINE that are destined for voice mail in accordance with UCR 2008, Section 5.2.2.3, Precedence Call Diversion.

6.1.6.8.11 Management Capabilities for Terminal Devices

[Required] The DCVX shall have the capability to manage its supported terminal devices as published in its HLR so it can assign, transfer, or terminate services, features, and calling capability to include telephone numbers for its terminal devices.

6.1.6.9 Terminal Device Specific Requirements

Cellular handsets often referred to as mobile subscribers, handsets, PDAs, “blackberries,” and any other user cellular end item devices commercial or Government developed, are herein referred to as terminal devices. The terminal device is the interface between the user and the cell network. The terminal device can be a handheld unit, a mounted mobile device, or a fixed location device.

6.1.6.9.1 Terminal Device Requirements

[Required] The terminal device shall provide the following status information to the network:

- Powered on
- Moved to a new location
- Alerting
- Dialing

[Required] The terminal device shall display the following status information to the end user:

- Signal strength
- Battery capacity
- Roaming status
- Service not available
- Call progress status

[Conditional] The terminal device may have the ability to provide key-locking ability to lock the terminal device's keypad and unlock the keypad after providing the appropriate key sequence/PIN number entries as provided by the vendor in the terminal device. The lock and unlock key sequence/PIN shall be settable by the user. There will be an administrator method that can be vendor proprietary, which can unlock the terminal device, in case the user PIN is not available or supplied.

[Conditional] The terminal device may have the capability to support WPS on commercial networks and/or DoD networks where provided when not connected to and functioning on a DoD precedence and preemption network.

[Required] Removable/Exchangeable SIM: The SIM card in commercially available terminal devices shall be removable and exchangeable into other similar commercially available terminal devices compatible with the DCVX system (applicable to a GSM-based system). This excludes secure terminal devices and other terminal devices not readily commercially available.

6.1.6.9.2 Terminal Device Signaling

[Required] The terminal device shall provide information to allow the DCVX to identify the terminal device when the terminal device is powered up, successfully registered, and in active call status.

6.1.6.9.3 Terminal Device Frequency Band Support

A terminal device that supports more than one frequency band has a high connection and reliability capacity.

[Conditional] A terminal device may support up to five frequency bands as specified in Table 6.1.6-1, Current Cellular Systems Parameters, for each protocol supported in Section 6.1.6.8.2, Protocol/Format.

[Conditional] The terminal device may also support roaming and interconnecting with commercial cellular networks when operating outside the transmission range of the home based DCVX and other supporting DCVXs interconnected in support of roaming within the Tactical OAN.

6.1.6.9.4 Terminal Device Encryption

[Required] If SCIP and/or other NSA-accredited encryption are implemented in the terminal device, the SCIP and/or other NSA-accredited encryption-capable terminal device shall have the capability to go secure; to provide E2E encryption to another secure cellular-capable terminal device; and via the DCVX, to a non-cellular NSA encryption-capable device per the requirements specified in Section 5.2.5 DoD Secure Communications Devices. The SCIP and/or other NSA-accredited encryption device shall provide E2E encryption within the DCVX, from DCVX to DCVX (roaming) and from DCVX to external networks such as DSN and/or PSTN.

[Conditional] The terminal device may support other non-NSA encryption schemas, such as AES encryption as used by the Government Emergency Telecommunications Service (GETS) system.

6.1.6.9.5 Terminal Device Battery Requirements

[Required] The readily commercially available non-secure terminal device must have a battery that shall provide as a minimum 6 days standby time in total and 3 hours non-secure talk time in total but not both requirements sequentially on the same battery charge. The NSA encryption secure terminal devices (e.g., PDA Secure Mobile Environment Portable Electronic Device (SME PED)) must provide their specified battery and secure/unsecure talk time. All other terminal devices must provide their specified battery and unsecure talk time and/or secure talk time, if applicable.

[Required] The terminal device shall have battery auxiliary capabilities when the primary battery is removed/drained to ensure primary network and user settings are not lost on the device before a primary battery is installed/recharged to ensure the terminal device is able to connect to the DCVX on power-up. Auxiliary battery shall provide a minimum of 2 hours of power to retain terminal device settings only.

6.1.6.9.6 Terminal Device Secure Call Handling

[Conditional] If the terminal device supports SCIP or other NSA-accredited encryption scheme(s), the terminal device/DCVX system will provide Classified secure call handling features, as defined in Section 6.1.4.8, Secure Call Handling.

6.1.6.9.7 Terminal Device Display/Alerting Features

The terminal device shall have the following display and alerting features:

1. **[Required]** Power-On Status: When the terminal device is powered on, the display shall indicate:

    a. Signal strength
    b. Remaining battery capacity
    c. Active call status
    d. HLR registration results (either success or failure)

2. **[Required]** Routine Call Alerting: The idle, registered terminal device shall provide or be provided an auditory and/or visual display alert for incoming routine calls.

3. **[Conditional]** Precedence Call Alerting: The DCVX may be required to meet the eMLPP functionalities specified in Section 6.1.6.6.3, Precedence and Preemption. The eMLPP references or uses a proprietary methodology. If precedence and preemption capability is provided, upon receiving a precedence call, the idle, registered terminal device will provide or be provided precedence alert and/or tone notification. Whether using eMLPP or proprietary version, the terminal device shall issue the same alerting tone(s) for precedence calls in accordance with eMLPP requirements. Upon notification, the user will have the capability to select or reject the call of higher precedence.

6.1.6.10 Base Station Subsystem Specific Requirements

The BSS minimally consists of the BSC, base station transceiver station(s), and the cell tower(s) with radio transceiver(s).

6.1.6.10.1 Signaling

[Required] The base transceiver for radio cells will determine which channel to use for call setup IAW the appropriate supported protocols listed in Section 6.1.6.8.2, Protocol/Format.

6.1.6.10.2 Strength

[Required] The base transceiver for radio cells will monitor the terminal device for signal strength and transfer the terminal device to the stronger cell when necessary IAW the appropriate supported protocols listed in Section 6.1.6.8.2, Protocol/Format.

6.1.6.10.3 Protocol/Format

[Required] The BSS shall support one or more of the protocols listed in the DCVX General Requirements, Section 6.1.6.8.2, Protocol/Format.

6.1.6.10.4 Coverage

[Required] For radio cellular, the BSC will assign the strongest cell to the terminal device. The coverage area this system will provide shall be in accordance with the GSM/CDMA standards and specifications in accordance with Table 6.1.6-1, Current Cellular Systems Parameters, and Section 6.1.6.8.2, Protocol/Format. Actual coverage will be depend on topology/man-made structures, and frequencies.

6.1.6.10.5 Preemption

[Conditional] If precedence and preemption capability is provided in the DCVX, then when preemption for reuse occurs, the BSS must disable the old call but maintain the channel assignment to the terminal device to allow the set up of the new call. In the event where there are no idle channels and a precedence call is received, the BSC will find the lowest precedence channel and preempt that channel to allow for the higher level precedence call to be completed.

6.1.6.11 DMSC Specific Requirements

The DMSC will minimally consist of the MSO, a VLR, AUC, and EIR. The HLR does not need to be a local component part of the DMSC but it will be necessary for the DMSC to access a home location register to determine the attributes of any terminal device. Whether the HLR is local with the DMSC or is remotely queried, the HLR is a component of the DCVX under test.

6.1.6.11.1 Visitor Location Register

[Required] The DMSC shall maintain a VLR to allow service to any active terminal device operating in the area being served by the DCVX that is registered with the HLR. The VLR knows which BSS is serving the active terminal device.

6.1.6.11.2 Home Location Register

[Required] The DMSC shall connect to a HLR to determine the attributes of the terminal device currently being served by the DCVX. The information provided by the HLR will tell the DMSC where the terminal device is located. The HLR will indicate the terminal device attributes and status. The information on the terminal device from the HLR is stored in the VLR. The HLR can be co-located with the DMSC or deployed remotely. The local HLR may be queried by vendor proprietary methodology. The remote HLR can be queried using DMSC network trunk interfaces of CCS7, MLPP PRI, or ISDN PRI**.** Additionally, the local and/or remote HLR may be queried using CCS7 over IP (SIGTRAN), or AS-SIP.

[Required] HLR Storage: The HLR must store and support information on each terminal device registered to the network the HLR serves.

[Required] HLR Change and Propagation: The HLR must support changes to the terminal device information. Once the HLR receives the supported change information the HLR, whether local or remote from the DMSC, has 3 minutes to propagate the change information to the VLR. If the DCVX supports roaming, the HLR change must also propagate to the querying VLRs.

[Conditional] Intra-DCVX Queries: If roaming capability is supported in the DCVX, the HLR must support queries from other DCVXs using specified protocol methods for obtaining terminal device information (e.g., CDMA- and GSM-based queries).

6.1.6.11.3 Equipment Identity Register

[Required] To validate terminal devices to prevent a compromised terminal device from connecting to the cellular switch and obtain services, an EIR capability must be provided and integrated to work in conjunction with the Terminal Device Authentication Center process (see Section 6.1.6.11.4, Terminal Device Authentication Center) to prevent compromising the DCVX.

6.1.6.11.4 Terminal Device Authentication Center

[Required] To authenticate terminal devices as valid terminal devices associated with the DCVX, the cellular switch will use standard cellular techniques, industry best practices, and/or vendor proprietary processes integrated into the switch.

[Conditional] Terminal devices not assigned to the supporting DMSC HLR (e.g., roaming terminal devices) may be supported for authentication via the industry standard(s) and/or industry best practices for roaming authentication.

6.1.6.11.5 Mobile Switching Office Functions and Features

6.1.6.11.5.1 MSO MLPP Trunks and Interfaces

[Required] The MSO shall support one or more of the following TDM and/or IP trunks and interfaces, but may not connect to the DSN network with both types (TDM, or IP) simultaneously.

6.1.6.11.5.1.1 TDM Support

[Conditional] If TDM trunks are supported, then the following requirements apply as directed:

1. **[Required]** The MSO will minimally support ISDN PRI (T1/E1) as defined in UCR 2008, Section 5.2, Table 5.2-1, Trunk Types and Signaling Used in the DSN (including legacy interfaces) for trunks that connect to the DSN/PSTN without MLPP capability.

2. **[Conditional]** If precedence and preemption capability is provided in the DCVX, the MSO will support one or more of the following:

    a. MLPP PRI (ANSI T1.619a, and ITU Q.955.3 and Q.735.3) EO Access trunk as a minimum requirement. The MLPP PRI protocol will conform to the requirements for DSN trunks as defined in UCR 2008, Section 5.2, Table 5.2-1, Trunk Types and Signaling Used in the DSN, for trunks that connect to the DSN with MLPP capability.

    b. CCS7 for signaling and associated T1/E1 bearer trunks. The CCS7 shall be in accordance with the SS7 requirements specified in the most current ANSI T1.100 series of standards and shall be capable of internetworking with ITU-T SS7 networks. Exceptions to these standards are explicitly noted in CCS7 requirements as listed in UCR 2008, Section 5.2.4.6. Only those CCS7 requirements that differ from their corresponding ANSI common channel signaling standard section are included in UCR 2008, Section 5.2.4.6. CCS7 shall only connect on DoD networks, not to the PSTN and/or other non-Government networks.

3. **[Conditional]** The MSO may support TDM/CAS trunk IAW UCR 2008, Section 5.2, Table 5.2-1, Trunk Types and Signaling Used in the DSN, for EO Access Trunks.

4. **[Conditional]** The MSO may support CCS7 for signaling and associated T1/E1 bearer trunks. The CCS7 shall be in accordance with the SS7 requirements specified in the most current ANSI T1.100 series of standards and shall be capable of internetworking with ITU-T SS7 networks. Exceptions to these standards are explicitly noted in CCS7 requirements as listed in UCR 2008, Section 5.2.4.6. Only those CCS7 requirements that differ from their corresponding ANSI common channel signaling standard section are included in UCR 2008, Section 5.2.4.6. CCS7 shall only connect on DoD networks, not to the PSTN and/or other non-Government networks.

6.1.6.11.5.1.2 IP Trunking AS-SIP Support

[Conditional] If AS-SIP IP trunks are supported, then the DCVX shall comply with the stated requirements of an LSC, and if required, act as a SIP B2BUA for the terminal devices to meet the EI requirements in Section 5.3.2.

6.1.6.11.5.1.3 SIGTRAN

[Conditional] The MSO may support CCS7 over IP using SIGTRAN in accordance with IETF RFC 2719, architectural framework for signaling transport and other associated supporting RFCs. SIGTRAN shall be used only in connecting DCVXs together on DoD IP Networks within the Tactical OAN in support of roaming capability and/or querying the local/remote HLR. SIGTAN shall not connect to the PSTN and/or other non-Government networks.

6.1.6.11.5.2 Non-MLPP Networks Support

[Conditional] The MSO may support ISDN PRI (T1/E1) non-MLPP trunk for connecting to the PSTN and/or other non-Government networks. The ISDN PRI protocol will conform to the requirements for commercial trunks as defined in UCR 2008, Section 5.2, Table 5.2-1, Trunk Types and Signaling Used in the DSN, for trunks that connect to non-DSN networks.

6.1.6.11.5.3 Call Handling

[Required] The MSO shall handle both intraswitch calls and calls to and from the DSN, while recognizing a powered on terminal device that comes into its area. The MSO shall also receive information on the terminal device from the HLR and store that information in the VLR. Secure call handling shall be as stated in Section 6.1.4.8, Security.

6.1.6.12 Security

[Required] All components of the cellular system(s) shall meet security requirements, for each supported mode, as outlined in DODI 8510.01 and the applicable STIG.

6.1.6.13 DCVX Network Traffic Management Operating System

[Required] The DCVX switching systems shall provide NM data to the administering NM console via one of the three following physical interfaces:

- Ethernet/ TCP/IP (IEEE 802.3)
- Serial (RS-232)/Asynchronous
- Serial/Synchronous (X.25 and/or BX.25 variant)

All NM data, and configurable features and functions that are collected shall be, as a minimum, accessible through one of these interfaces. The DCVX must provide four separate data channels. They may be physically separate (e.g., four distinct physical interface points) or logically separate (e.g., four user sessions through a single Ethernet interface). The data shall be transmitted to the NM console in a way to allow for storage, recovery, and transfer of the information to or from digital media storage, and to allow for printing and/or screen display using methods, such as ASCII, binary, or hexadecimal data.

The data channels shall be used for and must be capable of providing:

- Alarm/log data
- Performance data (e.g., traffic data)
- Accounting data (e.g., CDR)
- Switch access (to perform switch datafill administration and network controls)

6.1.6.14 Submission of Wireless Systems to UCCO for DSN Connection Request

[Required] The DCVX systems shall be engineered properly so that the BSS and cellular terminal devices achieve the required performance requirements in their specific deployed environment. The user shall submit a network design and engineering performance analysis with supporting calculations to meet minimum MOS performance with the request for DSN connection. For certification procedures, the UCCO submittal shall include wireless security compliancy as identified in Section 6.1.6.12, Security.

6.1.7 Deployed Tactical Radio Requirements

The requirements discussed in this section refer to post-2012 system deployments. This section does not discuss transition between current system deployments and the systems described herein.

6.1.7.1 Introduction and Purpose

The following sections describe the requirements that shall be met by all deployed TRNs for them to be certified and used in the OAN tier of the GIG. Requirements are defined at the system level as well as the various components that make up the radio networks, including protocol requirements. Several of these requirements reflect changes described elsewhere in this UCR. These will be indicated in the text.

The scope of this section is limited to push-to-talk (PTT) TRNs. Future updates will address radios that have the ability to dial directly to a DISN VoIP EI.

6.1.7.2 Applicability

The requirements within this section are applicable to all PTT-based TRNs that connect directly or indirectly to the DISN VoIP services.

The current version of the UCR~~-~~ is the governing requirements document that takes precedence over the explicit or implicit requirements of subsidiary or reference documents, standards, and specifications, ~~–~~except for those requirements specified in the documents listed in Section 6.1.6.3, Policy and Reference Documents.

6.1.7.3 Policy and Reference Documents

The policy and instruction documents in Section 6.1.6.3 will, in conjunction with the UCR~~ 2~~, be used as a basis for APL certification.

6.1.7.4 TRN System Overview

The TRNs provide wireless communication services with military-unique features. They differ from commercial, standards-based cellular networks in that individual radios within the TRN can communicate with each other, without the need for a base station, BSC, or constituent signaling or interconnect equipment. The TRNs use multicast-based RF transmissions to enable a set of radios using the same frequencies to communicate with each other.

The lower portion of Figure 6.1.7-1, TRN Connectivity, shows the architecture for a notional TRN and how the TRN connects to the DISN UC-compliant service. This connection is facilitated by a new UCR function, called the Radio Bridge Function (RBF). The RBF is a component of a deployed LSC, or a component of a radio within the TRN.

The upper portion of Figure 6.1.7-1 shows elements of the UC-compliant network and interconnects, as described elsewhere in this UCR.

The TRN, also known as a "Voice Net" for this description, is composed of Voice Net Segments. Each Voice Net Segment is a group of radios, which communicate on a common set of frequencies. At least one radio in each Voice Net Segment is designated as a Voice Net Access Radio (VNAR).

[Conditional] A VNAR performs as many as three roles, depending on the type of Voice Net. It acts as a conventional radio to communicate with other radios in its Voice Net Segment. If there is more than one Voice Net Segment in a Voice Net, the VNAR shall communicate with VNARs in the other Voice Net Segments using, what could be, a proprietary, packet-based radio access network (RAN). At least one VNAR in a Voice Net may also act as a UCR-compliant (e.g., APL-listed) EI to enable voice communications between the Voice Net and UCR-compliant VoIP EIs.

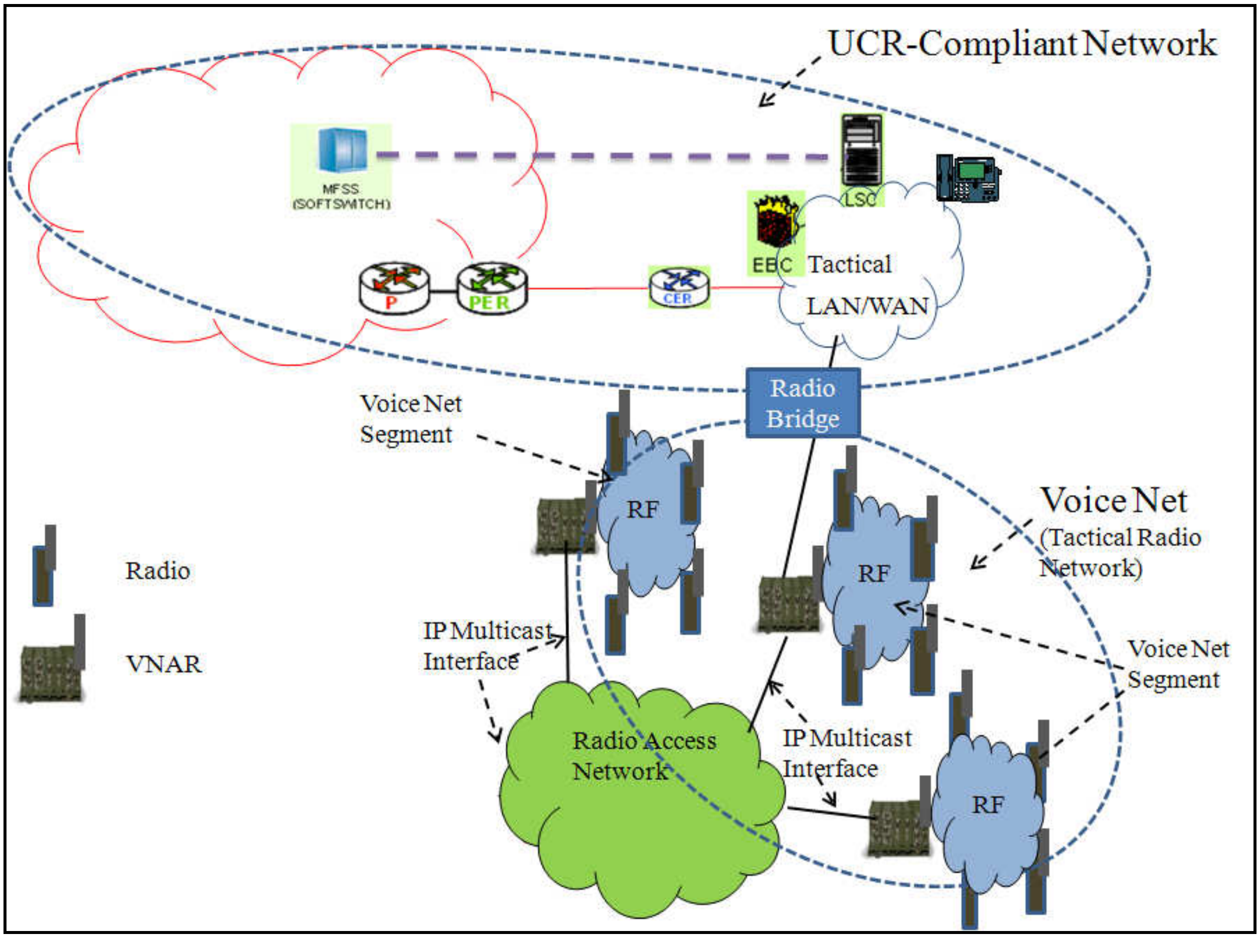


Figure 6.1.7-1. TRN Connectivity

The methods and formats for VNAR communications over the RF links and the RAN depend on the type of technology employed to create the Voice Net. Some Voice Nets operate in multicast PTT mode, where one party speaks and the others listen. Access to the radio links is controlled by layer 2 access methods used by all radios in the Voice Net Segment, and by human protocols. Communications is half-duplex by design or by enforcement of human protocol. Other Voice Nets support point-to-point communications initiated by one radio connecting to one or a few designated radios using full duplex communications. This version of the UCR is limited to describing requirements for the support of PTT-based voice networks.

Voice Net technology is not standardized. It is not the purpose of this UCR to create such standards. The requirements in this UCR are directed toward the communications between a VNAR and a UC-compliant voice EI. The basic requirement is that a VNAR provide a standardized EI interface so that other EIs can connect to the VNAR and become parties to the Voice Net. Figure 6.1.7-1, TRN Connectivity, shows the VNAR connected directly to an ASLAN. However, the VNAR could also connect via the RAN to a device that connects to an ASLAN.

6.1.7.5 Functional Description

This section defines the requirements for VNAR UCR compliance and requirements changes to enable Strategic and deployed LSCs to support traffic flow between UCR-compliant EIs and VNARs.

Figure 6.1.7-2, Functional Connectivity, provides an overview of the major functions necessary to provide connectivity between VoIP EIs and a TRN. This figure does not include the specifics of how the functions connect to each other.

The flow, shown in Figure 6.1.7-2, assumes that PTT signaling will be sent out-of-band, via the LSCs using the concepts defined in RFC 4730. Alternatively, the PTT signaling could be sent in-band, using the concepts defined in RFC 4733. In such case, the tone signals will be sent along the bearer path between the EI (or a proxy for the EI) and the RBF.

The EIs and LSCs connect over a UCR-compliant IP network, as indicated in Figure 6.1.7-1. The EIs and LSCs perform the functions defined for these elements as described within this UCR (see Section 4.4.1.1.2.2). In addition, the EIs and LSCs shall be enhanced to support a new set of AS-SIP signals that support PTT requirements for Tactical radios. LSCa and LSCb in Figure 6.1.7-2 support fixed EIs. The deployed LSC (DLSC) in the figure supports deployed TRNs and cellular networks as described in Section 6.1.6, Deployed Cellular Voice Exchange System Requirements, and for DLSC (TBD).

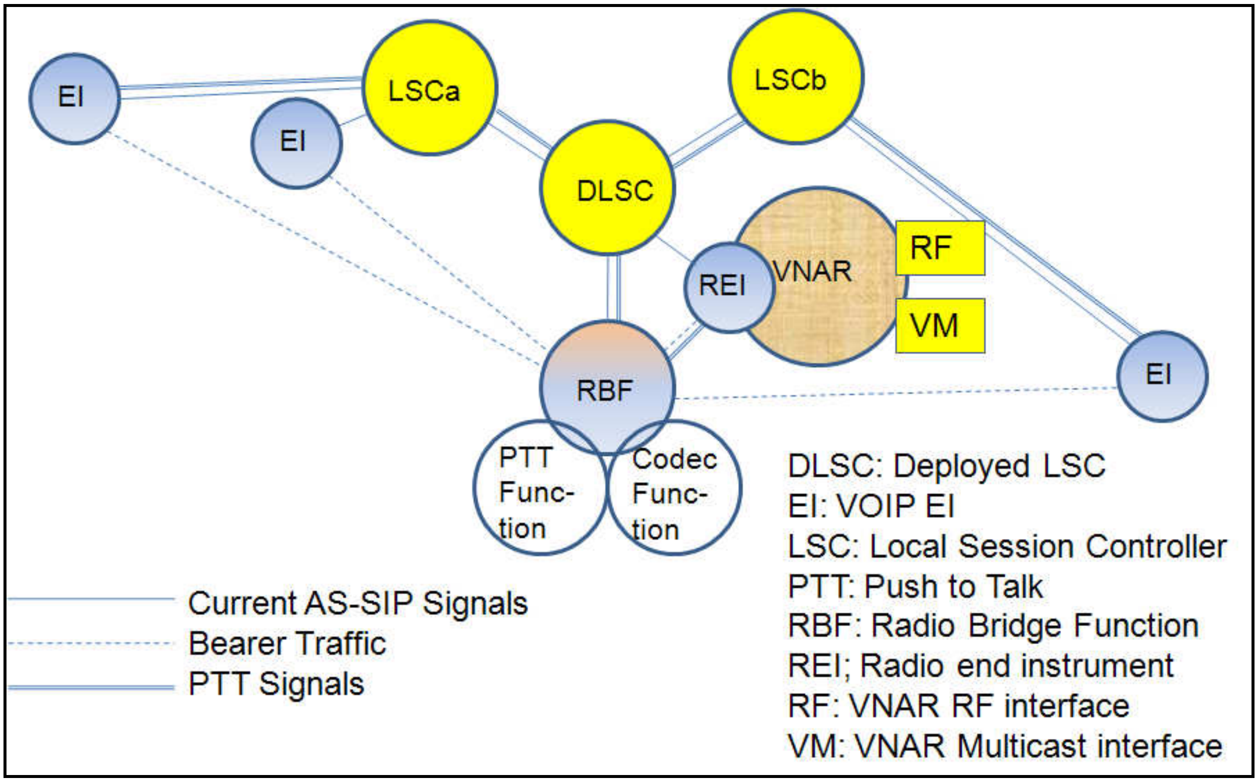


Figure 6.1.7-2. Functional Connectivity

The RBF provides connectivity between the EIs and the Radio End Instrument (REI). It is an enhanced version of a conventional conference bridge described in UCR 2008, Section 5.2.1.6.2.

The REI provides connectivity between the UCR-compliant domain and the Voice Net. The connection could require multiple router and switch hops. The RBF and REI also connect to the DISN IP voice network. The connection could be a LAN, WAN, or a back plane (if the functions are collocated in the same physical device).

The RBF could be a standalone appliance, or incorporated in a DLSC or incorporated in a VNAR. The PTT function, associated with the RBF, emulates the PTT function of the TRN to enable a conference participant to access the Voice Net. The codec function associated with the RBF performs transcoding, as necessary, to enable media transfer between the REI and the EIs. The REI is always located within a VNAR. The VNAR is connected to an RF element, and possibly a RAN.

The EIs and REIs join a conference, which is dedicated to the Voice Net. The RBF provides half-duplex, PTT access for the EIs to the Voice Net and enables a conference participant to speak to the Voice Net. AS-SIP is enhanced to provide a new set of signaling functions that support PTT access as described in Table 6.1.7-1, Control Information: VNAR to VoIP EI, and Table 6.1.7-2, Control Information: VoIP EI to VNAR.

Voice Net bearer traffic, shown as dotted lines in Figure 6.1.7-2, flows between the REI and the RBF where it is replicated for transmission to each EI that has been admitted to the conference. Bearer traffic flows from each EI to the RBF. In general, this traffic is blocked at the RBF. The only exception is bearer traffic from an EI that has been granted access via the PTT function.

Current AS-SIP signaling traffic is shown as solid lines in Figure 6.1.7-2. The PTT signaling traffic is shown as double lines in Figure 6.1.7-2. Both types of signal traffic flow between an EI and its Master LSC, from LSC to DLSC (possibly via multifunction softswitches (MFSSs), which are not shown in the figure) to the RBF and the REI and in the reverse direction. The PTT signaling traffic also flows in both directions between the RBF and the REI.

6.1.7.5.1 Radio Bridge Function

1. **[Required: RBF]** The system shall support three types of participants in a “Meet-me Bridge” Voice Net Conference (see UCR 2008, Section 5.2.1.6.2):

    a. Conference participant – An individual who joins the Voice Net using VoIP EIs.

    b. VNAR participant – One or more VNARs connected to the Voice Net.

    c. Conference manager – A conference participant who has the authority to manage certain features of the conference.

    Each participant joins the conference by calling in to a unique telephone number or URI that is assigned to the conference. The RBF will be assigned a unique IP address that corresponds to its telephone number and URI.

2. **[Required: RBF]** The system shall lead the caller through an authentication process using voice messages. If the process determines the caller is authorized join the conference, the system shall send a voice message informing the caller he/she is now a conference participant and indicate the status of the conference. Status shall include:

    a. Conference is not yet available.

    b. Conference has been terminated.

    c. Conference is in process including the number of participants, and an indication if the VNAR is a participant.

3. **[Required: RBF]** If the authentication process determines that the caller is not authorized, the system shall send a voice message informing the caller that he/she is not eligible for the conference. If the caller does not hang up in a parameter-determined period

of time, the system will terminate the call, whereas the parameter termination period shall be determined using a configurable time-out parameter with a time-out range of 0–60 seconds; default shall be set to 5 seconds.

4. **[Required: RBF]** The authentication process shall include a participant code, which identifies the type of participant, and a log-in process suitable to the security level of the conference.

5. **[Required: RBF]** The system shall perform the following functions in addition to those described in UCR 2008, Section 5.2.1.6.2:

   a. In the default mode a conference participant is placed in a listen-only mode, wherein the participant can only hear audio transmitted from the VNAR.

   b. The system performs whatever codec transformations are necessary to ensure compatible communications between the VNAR and the EIs (see Section 6.1.7.5.4)

   c. The system supports the PTT function defined in Section 6.1.7.5.3, Push-to-Talk Functional Requirements. The PTT function will ensure that only one conference participant can speak to the Voice Net at a time, and only when there is no other party speaking on the Voice Net (see Section 6.1.7.5.3).

   d. The conference manager has the ability to block or preempt any participant from access to the Voice Net.

   e. The conference manager has the ability to bridge the conference participants to each other, so they can speak with each other. The traffic resulting from this bridging will not be transmitted to the Voice Net.

   f. The conference manager has the ability to speak to any or all of the conference participants.

   g. The conference manager has the ability to terminate the speaker/listener status of any conference or VNAR participant.

6. **[Required: RBF]** The system shall operate as an AS-SIP EI for the purpose of authenticating, registering, and interacting with the LSC to originate or terminate voice sessions.

   NOTE: The RBF exchanges AS-SIP signaling packets with its Master DLSC. The DLSC exchanges AS-SIP messages with other LSCs or MFSSs to complete and tear down calls.

The Master DLSC could be collocated with or remote from the RBF. The Master DLSC could be uniquely assigned to the RBF, or could support multiple RBFs.

7. **[Required: RBF]** The system shall provide a method for multiple VoIP EI users to concurrently connect to the Voice Net, up to a configurable limit.

8. **[Required: RBF]** The system shall allow automatic termination of the session, based on configurable events, including inactivity on an AS-SIP session for a specified session time limit.

9. **[Required: RBF]** The system shall support MLPP requirements if more calls arrive than can be supported (see UCR 2008, Sections 5.2.2.1.2 and 5.2.12.8.2.3). The system shall be able to preempt a call from a lower precedence conference participant if necessary to provide resources to accept a call from a higher precedence conference participant if that call would otherwise be blocked. The system shall not preempt a call from a VNAR participant unless directed to do so by the conference manager.

10. **[Required: RBF]** The system shall support a configurable number of simultaneous conference participants per Voice Net.

11. **[Required: RBF]** The system shall support a configurable number of simultaneous VNAR participants per Voice Net.

12. **[Required: RBF]** The conference manager shall select one VNAR to act as master.

13. **[Required: RBF]** When the system is not acting as a master, the system shall act as a backup which is available to replace the master system if the master fails. It is highly desirable that the system be implemented to support an automatic fail-over to a backup VNAR if a master VNAR or its connection fails.

    NOTE: The number of VNAR and conference participants is not specified. These numbers are left up to best design and engineering practices as determined by the supplier of the RBF function to meet the performance and reliability goals required for a particular deployment.

6.1.7.5.2 Radio End Instrument

1. **[Required: REI]** The system, which resides within the VNAR, shall act as a conventional UCR-compliant EI in performing the following features defined in the following sections:

    a. Point-to-Point Call (Sections 5.3.4.13.2 and 5.3.4.19.2)

b. Tracing of terminating call (UCR 2008, Sections 5.2.1.4.2 and 5.3.2.2.2.2.2)

c. Outgoing call tracing (UCR 2008, Sections 5.2.1.4.3 and 5.3.2.2.2.2.3)

d. Tracing of a call in progress (UCR 2008, Sections 5.2.1.4.5 and 5.3.2.2.2.2.4)

2. **[Required: REI]** The system shall transform bearer traffic packets received from the RBF to the format associated with its Voice Net and transmit the traffic to the Voice Net.

3. **[Required: REI]** The system shall transform bearer traffic received from the Voice Net to bearer traffic packets for transmission to the RBF.

4. **[Required: REI]** The system shall transform UCR standard PTT signals defined in Section 6.1.7.5.3, Push-to-Talk Functional Requirements, to the PTT signals required by the VNAR to support the Voice Net.

5. **[Required: REI]** The system shall transform PTT signals received from the Voice Net to UCR standard PTT signals for transmission to the RBF

6. **[Required: REI]** The system shall operate as an AS-SIP EI for the purpose of authenticating, registering, and interacting with its Master DLSC to originate or terminate voice sessions.

   NOTE: The DLSC could be collocated with or remote from the REI. The DLSC could be uniquely assigned to the REI, or it could support multiple Voice Nets. The DLSC provides an MLPP function to give priority to higher precedence callers, if there is insufficient capacity to support a new dial-in call.

7. **[Required: REI]** The system shall support incoming session setup requests from IP EIs according the AS-SIP specification (Reference: "Department of Defense Assured Service Session Initiation Protocol (AS-SIP) Generic System Requirement (GSR)," DISA, Version 1.2.1, 12 May 2006).

8. **[Conditional: REI]** The system shall support a call to a VoIP EI directly without going through an external RFB. If this option is invoked the REI shall support:

   a. The PTT function described in Section 6.1.7.5.3, Push-to-Talk Functional Requirements

   b. The codec translation function described in Section 6.1.7.5.4, Codec Translation Function

c. MLPP functions described in UCR 2008, Section 5.2.2

d. Three-way calling as described in UCR 2008, Section 5.2.2.8.5

6.1.7.5.3 Push-to-Talk Functional Requirements

The VNAR, RBF, LSCs, and REIs cooperate to provide a capability that will enable a VoIP end user to initiate and terminate the equivalent of a PTT session.

1. **[Required: RBF]** The system shall provide a fail-safe mechanism to prevent a VoIP EI from streaming continuous voice traffic to a PTT-based Voice Net.

2. **[Required: RBF]** The system fail-safe mechanism shall ensure that, no matter the status of the VoIP end user or the VoIP EI, transmissions from the VoIP EI will terminate within a configurable, parameter-driven amount of time

3. **[Required: RBF]** The system fail-safe mechanism shall only reinstate transmissions based on upon completion of a specific, positive action by the VoIP end user.

4. **[Required: VNAR, RBF, REI – Conditional: LSC or DLSC]** The cooperating elements in a PTT session shall support DTMF tones after a call has progressed to the media session mode.

5. **[Required: RBF]** If PTT is configured, the system shall prevent bearer traffic generated from a VoIP EI from accessing the Voice Net until the VoIP end user initiates a PTT session by entering a unique configurable tone sequence called the "Talk Tone." This action mimics depression of the PTT button on a radio, thereby initiating an emulated PTT session. The VoIP end user enters a different, configurable tone sequence ("End Tone") to end the PTT session. This action emulates the release of the PTT button.

6. **[Required: RBF]** Upon receipt of the Talk Tone, the system in cooperation with the REI, shall determine whether the Voice Net is busy or available. The Voice Net is busy if any other party has been granted authorization to speak on the Voice Net. This can happen in one of two ways: 1) another conference participant has access to the Voice Net, or 2) there is a radio user has access to the Voice Net.

The following presents examples of traffic flow for the two cases where the Voice Net is busy, and the case where the Voice Net is available. These examples are representative, but not exhaustive. Traffic flows could vary based on the type of technology used in the Voice Net. The examples assume out-of-band signaling and the use of VNARs that can provide tone responses indicating a Voice Net available condition. The flow of traffic for the first busy case is as follows:

1. The participant keys in the Talk Tone sequence.

2. The Talk Tone is sent from the originating EI to its Master LSC.

3. The LSC converts the EI-generated Talk Tone to the UCR standard Talk Tone packet.

4. The LSC sends the Talk Tone packet to the DLSC to which the RBF is registered.

5. The RBF's Master DLSC sends the Talk Tone packet to the RBF.

6. In this case, the RBF determines that another conference participant has access to the Voice Net and cannot be pre-empted.

7. The RBF generates a standard "Busy Tone" packet for transmission to its Master DLSC.

8. That DLSC sends the Busy Tone packet to the LSC to which the requesting EI is registered.

9. The EI's Master LSC converts the tone to a form that is supported by the requesting EI.

10. The requesting EI creates an audio signal indicating that the Voice Net is busy.

The flow of traffic for the second busy case is as follows:

1. The participant keys in the Talk Tone sequence.

2. The Talk Tone is sent from the originating EI to its Master LSC.

3. The LSC converts the EI-generated Talk Tone to the UCR standard Talk Tone packet.

4. The LSC sends the Talk Tone packet to the DLSC to which the RBF is registered.

5. The Master RBF sends the Talk Tone packet to the RBF.

    a. In this case, the RBF determines that no other conference participant has access to the Voice Net.

    b. The RBF sends the Talk Tone packet to the REI.

    c. The REI translates the information in the packet to the form required by the VNAR to access the Voice Net. In this case, the Voice Net sends a signal back to the VNAR indicating that the Voice Net is busy.

d. The REI, within the VNAR, generates a standard "Busy Tone" packet and sends it to the RBF.

e. The RBF sets a flag indicating that the Voice Net is busy and sends the standard Busy Tone Packet its Master DLSC.

f. That DLSC sends the Busy Tone packet to the EI's Master LSC.

g. That LSC converts the tone to a form that is supported by the requesting EI, and sends the tone to the EI.

h. The requesting EI creates an audio signal indicating that the VNET is busy.

There is a timing issue in the busy cases. In some situations, the delay between the time of the initial VoIP EI PTT request and the time the request arrives at the VNAR could be in the 1–2 second range. This could lead to a situation where speakers on the TRN could block out VoIP speakers. To mitigate this situation, as an optional feature, the REI could store a blocked request from a VoIP EI, wait until the Voice Net is available, and then initiate the request to the Voice Net. In such case, the REI would send a busy tone immediately followed by an available tone to the EI, followed by periodic busy tones. When the Voice Net becomes available to the EI, the REI will send a burst of two available tones to the EI.

If the Voice Net is not busy, it is considered available. The signaling sequence for an available Voice Net, assuming out-of-band signaling is as follows:

1. The participant keys in the Talk Tone sequence.

2. The Talk Tone is sent from the originating EI to its Master LSC.

3. The Master LSC converts the Talk Tone to the UCR-standard Talk Tone packet.

4. The Master LSC sends the Talk Tone packet to the DLSC to which the RBF is registered.

5. The RBF's Master DLSC sends the Talk Tone packet to the RBF.

6. In this case, the RBF determines that no other conference participant has access to the Voice Net.

7. The RBF sends the Talk Tone packet to the REI.

8. The REI translates the information in the packet to the form required by the VNAR to access the Voice Net.

9. The REI translates the information in the packet to the form required by the VNAR to access the Voice Net. In this case, the Voice Net sends a signal back to the VNAR indicating that the Voice Net is available.

10. The REI, within the VNAR, formats a standard "Voice Net Available" packet and sends it to the RBF.

11. The RBF performs the following functions:

    a. Sets a flag indicating that the Voice Net is busy

    b. Sends the Voice Net Available packet to its Master DLSC

    c. Starts a PTT timer based on a configurable parameter

    d. Enables voice traffic from the originating EI to flow to the REI

    e. Blocks traffic from the Voice Net to the originating EI

12. The DLSC sends the Voice Net Available packet to the requesting EI's Master LSC.

13. That LSC converts the tone to a form that is supported by the requesting EI, and transmits the tone to the EI. The LSC also signals the EI to return to the media transmission mode.

14. The requesting EI creates an audio signal indicating that the Voice Net is available, and reverts to the media mode of operation.

15. The conference participant can now speak. Bearer traffic will be sent from the EI to the RBF. The RBF will send the bearer packets to the conference participants and the REI. The REI will translate the bearer traffic so that the VNAR can transmit the bearer traffic to the Voice Net.

16. **[Required: VNAR, RBF, REI – Conditional: LSC or DLSC]** The PTT session shall terminate upon any one of the following activities:

    a. The VoIP end user keys in the end tone. This tone is sent to the Master LSC and from there to the LSC serving the RBF and from there to the RBF.

    b. A voice activity detection (VAD) device in the RBF determines that there has been no voice activity for a configurable time-period.

    c. The PTT Timer expires

17. **[Required: RBF]** Upon termination of the PTT session, the system shall execute the following actions:

    a. Transmit a "PTT Terminated" tone packet to the VoIP EI (via the appropriate Master LSCs) for a configurable amount of time.

    b. Reset the PTT Timer.

    c. Reset the Voice Net busy flag.

    d. Re-enable the transmission of bearer traffic from the Voice Net to the VoIP EI.

    e. Send a PTT Terminated packet to the REI.

    f. The REI will convert the information in the PTT Terminated packet to the form required by the VNAR to terminate the PTT session in the Voice Net.

    g. The Voice Net will become available to other parties who wish to speak.

18. **[Required: RBF]** At some configurable time before a PTT time-out, the system shall issue a "Warning" packet to inform the speaker the session is about to terminate. The Warning packet will be transmitted via the signaling path, from the system's DLSC to the conference participant's LSC to the EI.

19. **[Required: RBF]** The system shall ignore a Talk Tone generated by a VoIP EI that is in a PTT session.

20. **[Required: RBF]** The system shall ignore an end tone generated by a VoIP EI that is not in a PTT session.

21. **[Objective: VoIP EI]** It is desirable if the VoIP EI could be modified to include a special control key that must be depressed to maintain the emulated PTT session. This would emulate the PTT action at a radio more accurately, and potentially reduce dead time associated with the use of a timer.

22. **[Required: RBF]** The system shall provide the following configurable mechanisms to mitigate situations where the VoIP EIs might not support tone patterns to define the beginning and end of a PTT session:

    a. The conference manager shall have the ability to place the VoIP EI in listen-only mode.

b. The system shall transmit a "Warning Tone" to the VoIP EI if there is voice traffic generated from a radio on the Voice Net.

c. The system shall invoke a configurable off-on feature, which will limit the time duration of transmissions from the VoIP EI. The system shall not forward traffic from such devices for more than the configurable amount of time. If the system terminates a PTT session based on this parameter, it shall not forward traffic from the VoIP EI until a configurable amount of time has passed since the end of the last transmission period.

23. **[Required: RBF]** Table 6.1.7-1, Control Information: VNAR to VoIP EI, defines the standard tones that shall be used to convey status information to the VoIP EIs.

Table 6.1.7-1. Control Information: VNAR to VoIP EI

CONTROL SIGNAL ID	SIGNAL NAME	SIGNAL CONFIGURATION (TONES)	DESCRIPTION	ACTION
1	Voice Net Available	TBD	The Voice Net is operational, and is not busy.	Caller can start to talk. Typically sent in response to a Talk Tone request sent by the caller (See Table 6.1.7-2).
2	Voice Net Operational But Not Available	TBD	The Voice Net is operational but cannot be accessed because lack of resources to support call; also used to indicated that the call has been preempted.	Caller should hang up and try again later. Typically sent in response to a Talk Tone.
3	Voice Net Busy – "Busy Tone"	TBD	The Voice Net is operational and reachable, but is busy.	Caller should try again later. Typically sent in response to a Talk Tone.
4	PTT Terminated	TBD	VNAR has terminated the PTT session based on a request from the VoIP caller or a timeout.	For information purposes; caller should stop talking.
5	Voice Net Secure	TBD	Transmissions from the VNAR to the Voice Net are sent in encrypted or scrambled mode.	For information purposes. Typically sent in response to a Talk Tone.

CONTROL SIGNAL ID	SIGNAL NAME	SIGNAL CONFIGURATION (TONES)	DESCRIPTION	ACTION
6	Voice Net Plain Text	TBD	Transmissions from the VNAR to Voice Net are sent in Plain Text mode.	Caller should not talk if he/she were expecting that transmissions would be secure. Typically sent in response to a Talk Tone.
7	Stop Transmitting	TBD	Stop talking: either the speaker is talking for too long a time, or there is a higher priority speaker who needs access to the Voice Net.	VNAR will block voice from reaching Voice Net. Caller should stop talking.
8	Warning	TBD	VNAR will terminate voice traffic from VoIP EI within a configurable period.	Caller should be aware that he/she will soon get a stop transmitting signal.

NOTE: The typical Voice Net will not generate all the control signals shown in Table 6.1.7-1. It is up to the designer of the VNAR to determine which control signals must be implemented. However, if a signal shown in Table 6.1.7-1 is used, the tones transmitted shall be the ones defined in Table 6.1.7-1. It is possible that future generations of TRNs will create additional signals. In such case, this UCR will be modified to accommodate the new signals.

Table 6.1.7-2, Control Information: VoIP EI to VNAR, defines the signals and standard tones that shall be used at the VoIP EI to indicate the start and termination of a PTT session.

Table 6.1.7-1 and 6.1.7-2 do not include AS-SIP signaling, which is discussed in Section 5.3.2.

Table 6.1.7-2. Control Information: VoIP EI to VNAR

CONTROL SIGNAL ID	SIGNAL NAME	SIGNAL CONFIGURATION (TONES)	DESCRIPTION	ACTION
1	PTT Request – "Talk Tone"	TBD	A tone that indicates the start of a PTT session.	VNAR will format and initiate PTT request to VNAR, provided that Voice Net has PTT capability.
2	PTT Terminate – "End Tone"	TBD	A tone that indicates the termination of a PTT session.	VNAR will terminate PTT request to Voice Net.

6.1.7.5.4 Codec Translation Functional

[Required: REI] The system shall support at least one of the following RTS VoIP codecs:

- G.711a and µ-law (64 kbps)
- G.723.1 (5.3 and 6.3 kbps)
- G.729d (6.4 kbps)
- G.729e (12.4 kbps)

[Required: REI] The system shall support the Enhanced Mixed Excitation Linear Production (MELPe) codec at 2.4 kbps and lower rates.

NOTE: When a call is set up between an RTS VoIP EI and the RBF, the codec function will negotiate the codec using the Session Description Protocol (SDP). When a call is set up between a REI and the RBF, the codec function will negotiate the codec using the SDP. The codec function will attempt to minimize the bandwidth required between the RBF and the REI.

[Required: RBF] The codec function is always associated with the RBF and will reside in the same device as the RBF.

6.1.7.5.5 End Instrument Functional Changes

[**Conditional: AEI, PEI]** The system shall support the PTT signaling functions described in Section 6.1.7.5.3, Push-to-Talk Functional Requirements.

[Objective] It is highly desirable that the EIs be configured to directly process the standard PTT signaling packets used to create the audio tones and convey user keystrokes during a PTT session

[Conditional] However, it is permissible to have the EI's Master LSC create the standard signaling packets and employ a non-standard approach to conveying that signaling information between the EI and its Master LSC.

[Objective] It is highly desirable that EI PTT signaling support be implemented by software and configuration downloads, rather than hardware changes in the EIs.

6.1.7.5.6 LSC Functional Changes

[Conditional: LSC] The system shall support the PTT signaling function additions to AS-SIP described in Section 6.1.7.5.3, Push-to-Talk Functional Requirements. If the EI does not directly create and process standard PTT signaling packets, it is highly desirable that the LSCs be able to download software changes to the EIs as described above in Section 6.1.7.5.5, End Instrument Functional Changes.

6.1.7.6 Network Management

General network management requirements are specified in UCR 2008, Section 5.2.8.

[Required: RBF] The system shall identify the unique name of the TRN supported by the conference.

6.1.7.7 Tactical LAN Requirements

The assured services objectives are difficult to achieve in the Tactical-edge networks (TNs), due to dynamically changing connectivity, limited bandwidth, unstable environment, and limited equipment. The TNs are often non-ASLAN compliant.

The UCR does not permit non-ASLAN compliant devices to support special C2 and C2 users. This architecture has to be modified to allow a EI located at a non-ASLAN Tactical location to support special C2 and C2 users so long as a type I encryption mechanism is applied to the call signaling messages and bearer traffic.

6.1.7.7.1 Physical Media Requirements

[Required: VNAR] The system must support at least one of the following Ethernet types:

- 10 Base-T
- 100 Base-T
- 1000 Base-T

6.1.7.7.2 Dial Plan & Routing Requirements

(Reference to Sections 5.3.2.16, UCR 2008 Sections 5.2.3.5 and 5.2.3.5.1)

[Required] Each Voice Net shall be assigned a routable user identity, which can be one of the following: DSN number, Tel-URI, SIP-URI, and FQDN.

6.1.7.7.3 DSCP

[Required: RBF, REI, VNAR] The system shall provide a configurable mechanism to mark DSCPs in the header of IP packets. However, the default marking shall be as defined in Section 5.3.3.

6.1.7.7.4 Traffic Engineering

The REI supports only voice and control traffic and can apportion that traffic in any manner as determined by traffic engineering. The number of subscribers needs to be supported will be determined by each Program Office (PO).

The VNAR should operate within the overall network voice E2E delay and jitter requirements in accordance with Section 5.3 voice requirements, specifically Section 5.3.1.4.1, Voice Services. The RECOMMENDED upper limit on the average post-selection delay for various E2E scenarios is defined in Table 6.1.7-3, Upper Limit on Average Post Selection Delay.

NOTE: The UCR and REI time delays stated in Table 6.1.7-3, Upper Limit on Average Post Selection Delay, relate only to the time it takes to set up a call to the RBF. This is typically done once per conference per VoIP EI and REI. The PTT signaling requests will occur many times during the conference. The time to implement a PTT request from a VoIP EI and return an available signal to the EI involves round-trip delay between EI and the REI. This time will vary from subseconds if there are no satellite links involved, to as many 2–3 seconds if satellite links are involved. The maximum delay to release an EI-initiated access to the TRN is determined by one way delay and is proportionately less.

Table 6.1.7-3. Upper Limit on Average Post Selection Delay

TYPE OF DELAY	UCR	REI
Local intratheater DSN call signaling during normal network traffic load	1 second	TBD
Local intrabase DSN call signaling during normal network traffic load	1.5 seconds	TBD
Worldwide DSN call signaling during normal network traffic load	6.0 seconds	TBD
Global DSN call signaling during normal network traffic load	8.0 seconds	TBD

6.1.8 UC Architecture for the Tactical Environment

This section provides requirements for LSCs and as such augments section 4.5.1.1.2.2 LSC Designs – Voice.

The section focuses on the deployed (tactical) use of the Master/Subtended LSC architecture and the introduction of Dynamic Assured Services Admission Control (DASAC). DASAC enables an LSC to admit, block or preempt new voice and video calls based on the communications capacity (bits/sec) required for the call and the link capacity available to support the call. DASAC will augment the current ASAC approach in which LSCs admit calls based on a call budget. DASAC will be applied independently to voice and video calls. DASAC details are fully specified in section 6.1.8.4.

[Required Deployed LSC- Master/Subtend Architecture Applies to all Deployed (Tactical) UC Real-time Services] The Master/Subtended LSC architecture supports all deployed UC real-time services including both voice and video service.

[Required Deployed LSC- Directionalization Budget Inheritance for All Real-time Including Voice and Video] DASAC inherits the voice directionalization ASAC budget requirements (i.e. IPB, IPBi, IPBo) from sections 5.3.2 Assured Services Requirements and 5.3.4 AS SIP Requirements and makes them available for all real-time session based UC services including voice and video.

Within deployed domains calls typically involve multiple bandwidth constrained links. Each such link must be subject to DASAC. These links are typically wireless (e.g. satellite, radio) in nature. Deployed sites generally exist within a tiered command and control hierarchy.

[Required Deployed LSC: Deployed LSCs and DASAC] Deployed LSCs must implement Dynamic Assured Service Admissions Control (DASAC).

6.1.8.1 Architectural Overview

The deployed site hierarchy is shown in Figure 6.1.8-1.

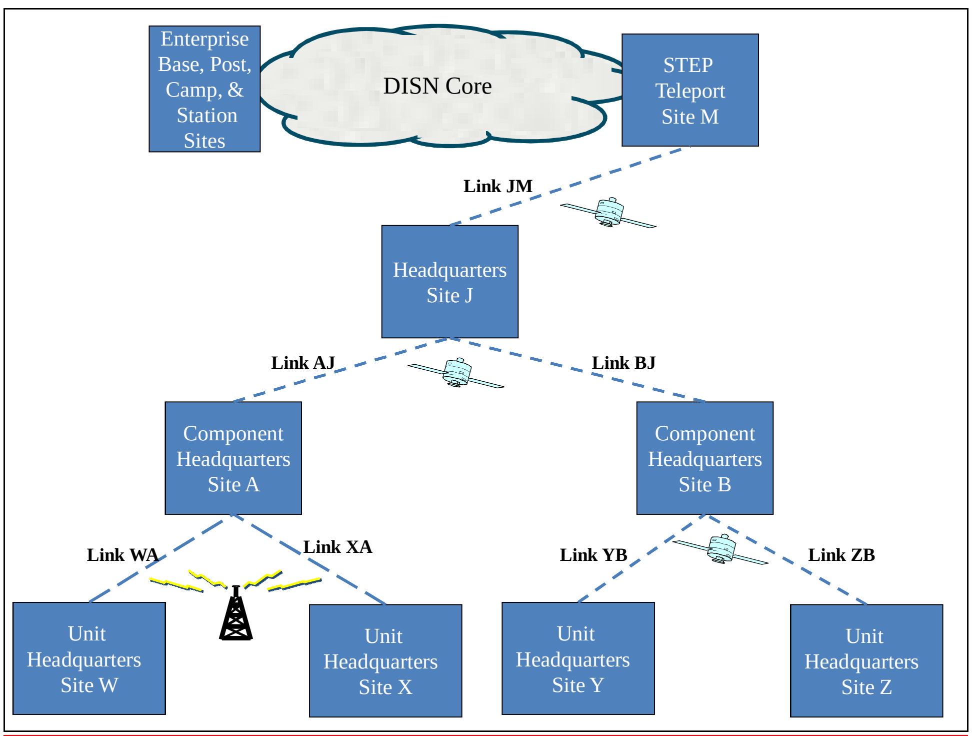


Figure 6.1.8-1 Deployed Hierarchy

DASAC budgets are required for each of the links WA, XA, YB, ZB, AJ, BJ, and JM in Figure 6.1.8-1. Links between component headquarter locations and headquarters locations support aggregated traffic. Further traffic aggregation occurs between headquarters and STEP sites. The Master/Subtended LSC architecture enables DASAC budgets to span multiple deployed links.

Each deployed link is bounded by a Master/Subtended LSC pair. The command and control hierarchy determines whether a LSC is "Master" or "Subtended". - For a given link the LSC in the "higher" site in the command and control hierarchy is designated the Master LSC. For the same link, the LSC in the "lower" site in the command and control hierarchy is designated the Subtended LSC. The Subtended LSC and its Master independently and in parallel apply their DASAC budgets to the shared link. The Master aggregates traffic from one or more subtended LSCs.

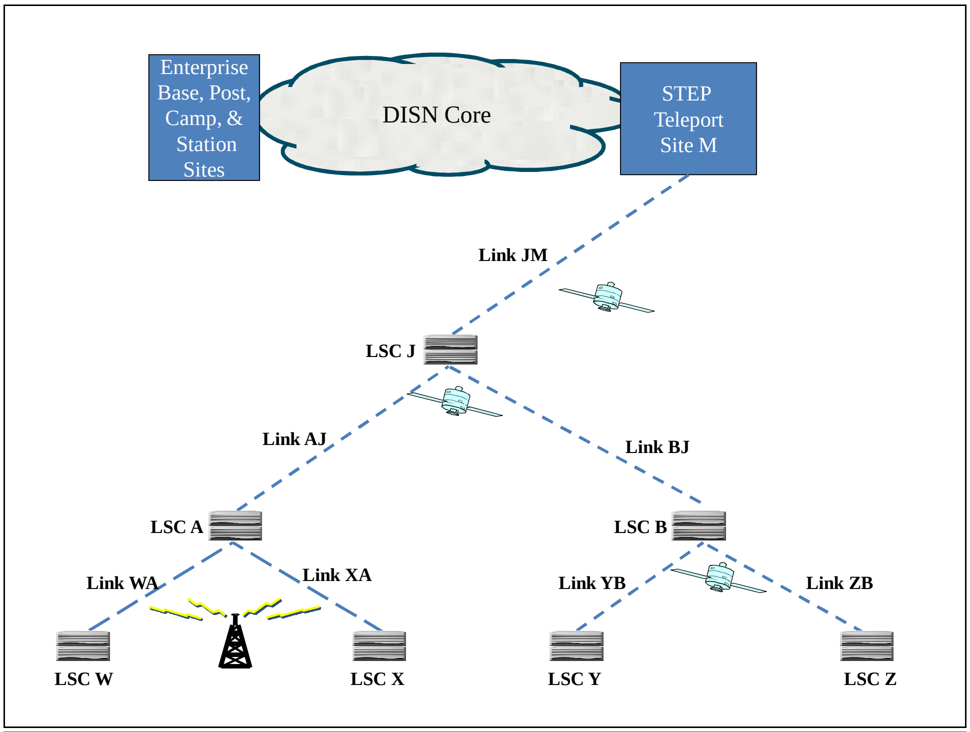


Figure 6.1.8-2 Deployed LSCs

Figure 6.1.8-2 shows deployed LSCs and their master/subtended relationship:

- LSCs W and A provide the DASAC audio and video budgets for Link WA. Subtended LSC W is subtended to the Master LSC A.
- LSCs X and A provide the DASAC audio and video budgets for Link XA. Subtended LSC X is subtended to the Master LSC A.
- LSCs Y and B provide the DASAC audio and video budgets for Link YB. Subtended LSC Y is subtended to the Master LSC B.

- LSCs Z and B provide the DASAC audio and video budgets for Link AB. Subtended LSC Z is subtended to the Master LSC B.

- LSCs A and J provide the DASAC audio and video budgets for Link AJ. Subtended LSC A is subtended to the Master LSC J. Recall that LSC A is also the Master LSC to Subtended LSCs W and X.

- LSCs B and J provide the DASAC audio and video budgets for Link BJ. Subtended LSC B is subtended to the Master LSC J. LSC B is also the Master LSC to Subtended LSCs Y and Z.

- LSC J provides the DASAC audio and video budgets for Link JM. LSC J is the Master LSC to Subtended LSCs A and B.

[Required Deployed LSC: Master/Subtended LSC Pair Administers Each Deployed Link] For every deployed link, except for links leading to the UC backbone, a Master/Subtended LSC pair will apply their respective DASAC budgets to the shared link. The Master LSC and the Subtended LSC DASAC budgets for a given media type for a specific link might differ.

In Figure 6.1.8-2 LSCs A and B play the role of both a master LSC and a subtended LSC as a function of the tiered command and control hierarchy. For reasons of scaling the Master/Subtended LSC architecture may also be employed within a deployed site.

[Required Deployed LSC: Deployed LSC Supports Multiple DASAC Budgets] A Deployed LSC must be able to support a DASAC budget by media type, voice or video, on an administered link by administered link basis.

[Required Deployed LSC: Master and Subtended LSCs Topology] A Subtended LSCs may be in the same or different deployed LAN as that of its Master LSC.

[Required Deployed LSC as Both a Master and a Subtended LSC] For different links a given LSC may serve as a Subtended LSC or a Master LSC.

[Required Deployed LSC: Master LSC and Its Primary and Secondary MFSS or WAN SS] A master LSC at the deployed/UC backbone edge will have both a primary MFSS or WAN SS and a secondary (backup) MFSS or WAN SS.

If a given LSC's end instrument supports SRTP and if there is no need for LSC-resident media gateway functionality, there is no need for the LSC to be involved in the end instrument's SRTP sessions. In the deployed environment the Master LSCs, Subtended LSCs, and EBCs are AS SIP back to back user agents. The EBCs also anchor the SRTP sessions. These sessions stretch from SRTP capable end instrument to SRTP end instrument; the EBCs do not terminate and re-originate the SRTP sessions. The EBCs relay the SRTP sessions and provide NAT translation support, if necessary.

[Required EBC, CER, Routers: Single SRTP Flow] Within a two party call for each media type in each direction there is only one SRTP flow.

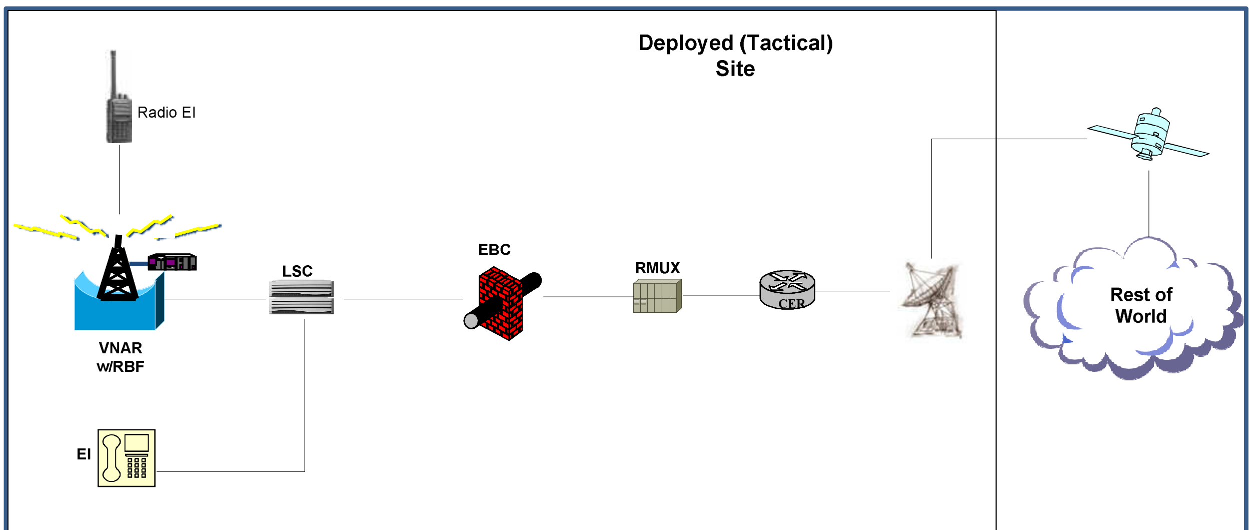
Every deployed LAN has an associated EBC. All packets marked for Assured Service in the deployed LAN must be delivered to this EBC. The EBC confirms that these packets have an associated AS SIP based session. If not, the packets do not receive Assured Service treatment.

[Required EBC: Deployed Sites and EBCs] Each deployed UC site must have an associated EBC.

[Required EBC, Routers: All UC Flows Anchored by the EBC] Within a deployed LAN all packets marked with the DSCP Assured Service values associated with the Granular Service Classes of Assured Voice and Assured Multimedia Conferencing per Tables 5.3.3-1 and 5.3.3-2 , must be routed to the EBC.

[Required EBC: All Assured Service Flows Policed by the EBC] If any packets marked with the DSCP Assured Service values do not have an association with an active AS SIP session, the EBC will police such calls to ensure that they do not collectively exceed a pre determined capacity requirement for the target link

Figure 6.1.8-3 depicts a conventional deployed LAN where all the UC elements reside together.



Legend:

EBC	Edge Boundary Controller	RBF	Radio Bridge Function
EI	End Instrument	RMUX	Real-time IP Mux
LSC	Local Session Controller	VNAR	Voice Net Access Radio

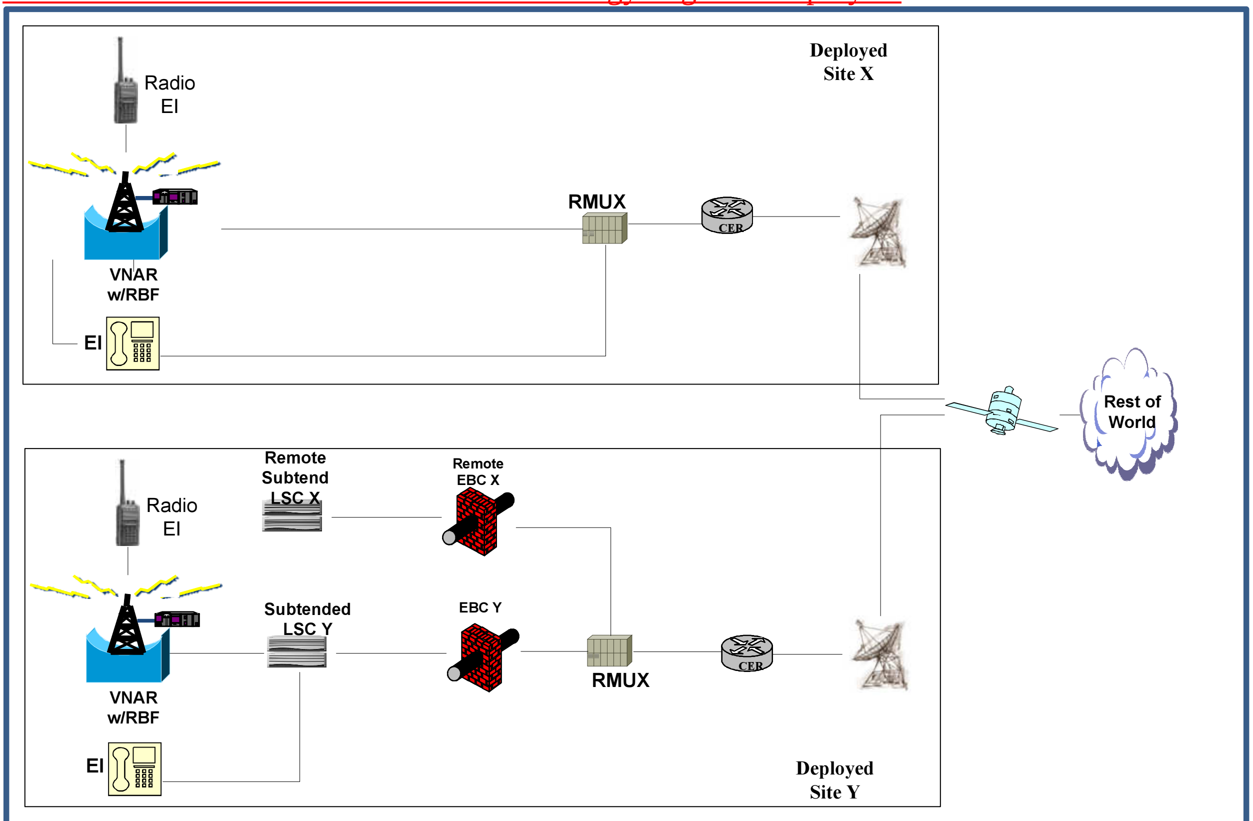
Figure 6.1.8-3 Deployed Site With All UC Elements

A deployed site's UC elements could be deployed remotely from the site. For example, Figure 6.1.8-4 is a depiction of Deployed site X's LSC and EBC assets physically residing at Deployed site Y. Remote elements enable deployed sites to reduce costs, footprint, weight, and power.

Remote placement also enables the movement critical UC elements to what might be a safer location relative to the battle zone.

If elements are deployed remotely, intra-site calls would rely upon these remote elements. Intra-site calls would incur additional setup delays and/or media delays. Also, intra-site calls would rely upon the availability and the capacity of wide area links (e.g. satellite links, radio links). The tradeoffs must be carefully considered before adopting such an approach.

An EBC or LSC could be deployed in several forms. The element might be shared and identified by a single IP address. The element might be partitioned, with each partition having its own IP address. Virtual machine middleware technology might be employed.



Legend

CER:	Customer Edge Router	RBF	Radio Bridge Function
EBC:	Edge Boundary Controller	RMUX	Real-time IP Mux
EI:	End Instrument	SLSC	Subtended LSC
MLSC: Master LSC		VNAR	Voice Net Access Radio

Figure 6.1.8-4 Highly Distributed Deployed Hierarchy

[Conditional LSC: Remote LSC] A deployed site's LSC may physically reside at a separate site.

[Conditional LSC: Logical LSC] A virtual LSC may house two or more logical LSCs supporting two or more deployed sites. Each logical LSC is a software based partition of the single physical LSC asset.   Each logical LSC will have its own IP address and its own CCA IDs. Virtual machine middleware may be employed for the partitioning of the physical LSC into two or more logical LSCs.

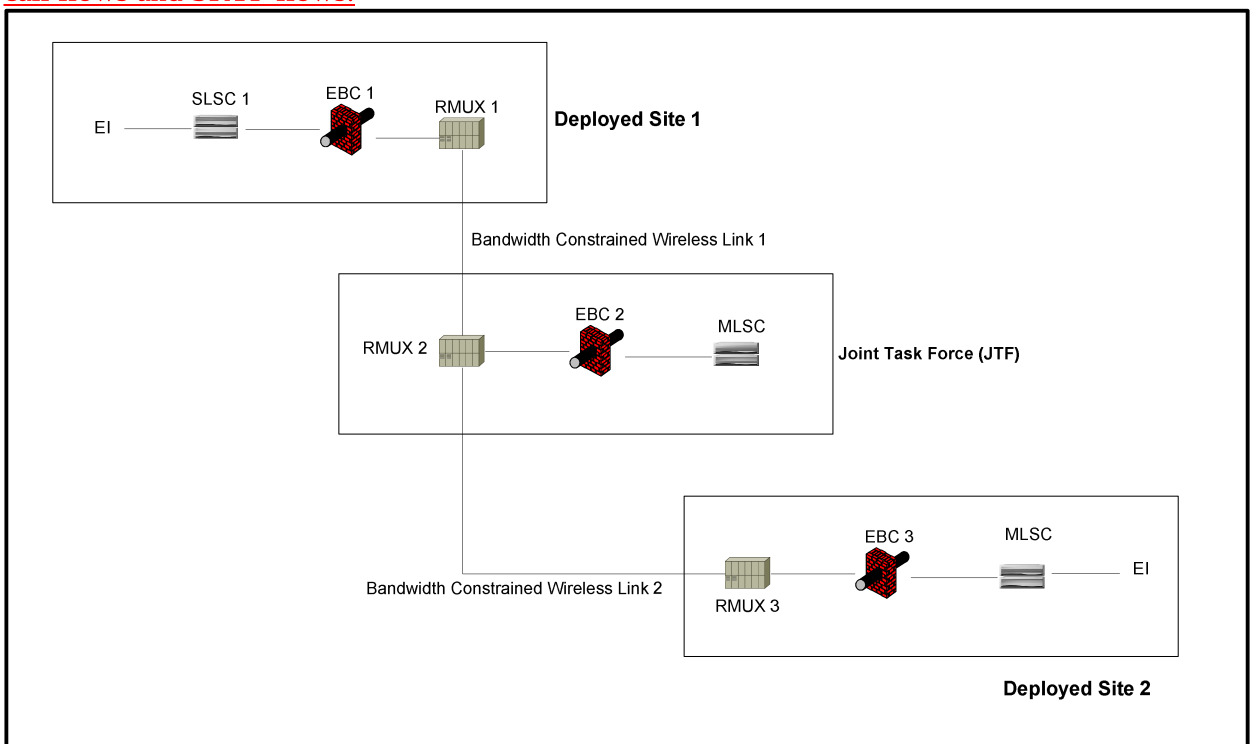
[Conditional EBC: Remote EBC] A deployed site's EBC may physically reside at a separate site.

[Conditional EBC: Shared EBC] An EBC may be shared with two or more sites.

[Conditional EBC: Logical EBC] A virtual EBC may house two or more logical EBCs supporting two or more deployed sites. Each logical EBC is a software based partition of the single physical EBC asset. -Each logical EBC will have its own IP address. Virtual machine middleware may be employed for the partitioning of the physical EBC into two or more logical EBCs.

6.1.8.2 Associated Sample Call Flows

Figure 6.1.8-5 depicts a LSC and an EBC at every deployed site. This simple UC network example will serve as the basis for several subsequent illustrative Master/Subtended LSC AS SIP call flows and SRTP flows.



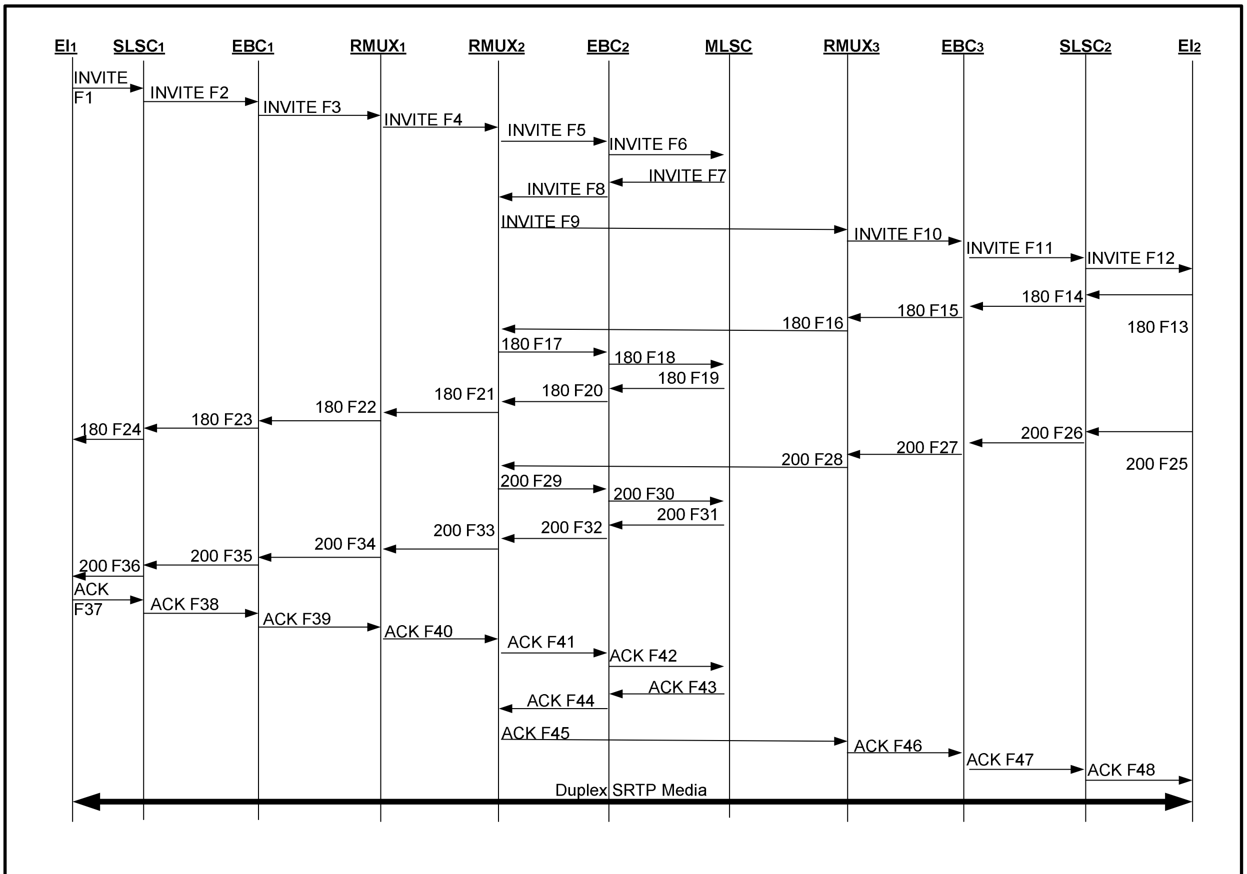
Legend:

EBC	Edge Boundary Controller	MLSC	Master LSC
EI	End Instrument	RMUX	Real-time IP Mux
SLSD	Subtended LSC		

Figure 6.1.8-5 Deployed (Tactical) Topology Examples

6.1.8.2.1 Basic Session Setup Deployed Site to Deployed Site Via JTF

Figure 6.1.8-6 shows the AS SIP message flow for call setup. Upon receipt of the AS SIP INVITE message SLCS 1 and the MLSC both perform DASAC processing for UC Assured Service calls on bandwidth constrained Link 1 shown in Figure 6.1.8-5. SLCS 2 and the MLSC both perform DASAC processing for UC Assured Service calls on bandwidth constrained Link 2 in Figure 6.1.8-5.



Legend:

ACK	AS SIP Message ACK	INVITE	AS SIP INVITE Message with SDP offer
BYE	AS SIP Message BYE	MLSC	Master LSC
EBC	Edge Boundary Controller	RMUX	Real-time IP Mux
EI	End Instrument	SLSC	Subtended LSC
180	AS SIP Message 180 Ringing	200	AS SIP Message 200 OK with SDP Answer

Figure 6.1.8-6 Basic Session Setup Deployed Site to Deployed Site Via JTF

6.1.8.2.2 Deployed to Deployed via JTF SRTP Flows

The SRTP flows are from end instrument to end instrument but they transit EBCs 1, 2, and 3 as well as the Real-time Muxes 1, 2, and 3. The flow is shown in Figure 6.1.8-7.

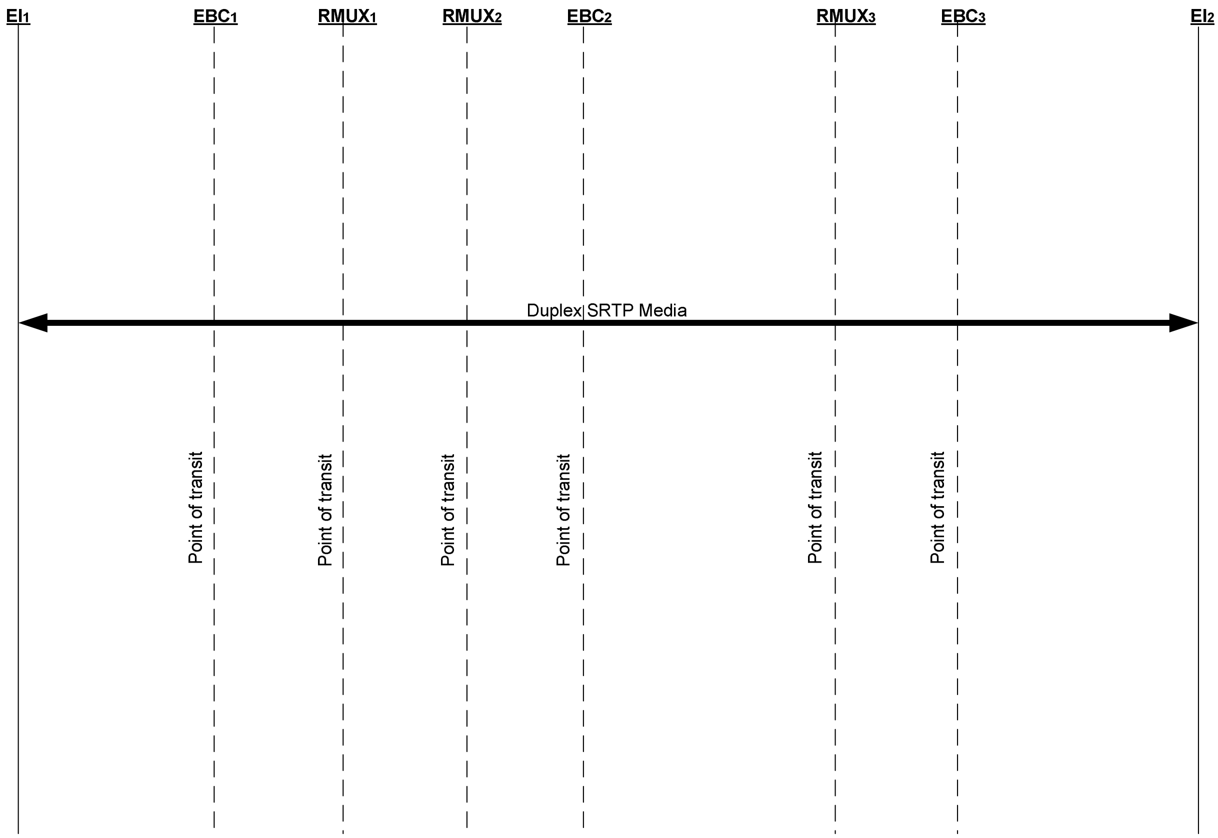


Figure 6.1.8-7 Deployed to Deployed via JTF SRTP Flows

6.1.8.2.3 Deployed to Deployed via JTF Session Teardown

AS SIP flow for session teardown is shown in Figure 6.1.8-8. Upon receipt of the AS SIP 200 OK message SLCS 2 and the MLSC both perform DASAC processing for UC Assured Service calls on bandwidth constrained Link 2 from diagram Figure 6.1.8-5. SLCS 1 and the MLSC both perform DASAC processing for UC Assured Service calls on bandwidth constrained Link 1.

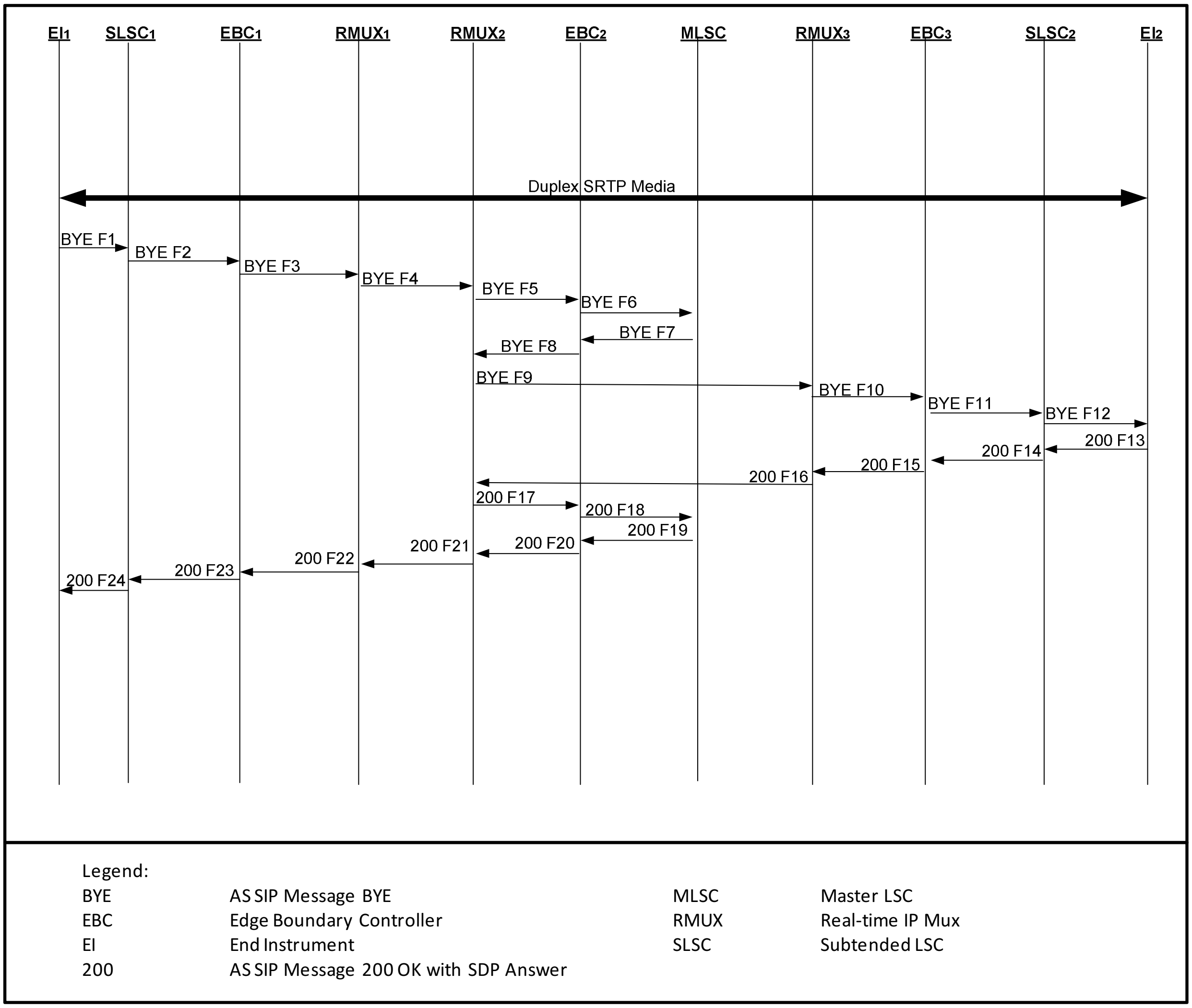


Figure 6.1.8-8 Deployed to Deployed via JTF Session Teardown

6.1.8.2.4 Basic Session Setup Deployed Site to Enterprise Via JTF

Figure 6.1.8-9 depicts a deployed (tactical) site to a fixed (strategic) enterprise site via the JTF and the UC backbone network. This simple UC network example will serve as the basis for several subsequent illustrative Master/Subtended LSC AS SIP call flows and SRTP flows.

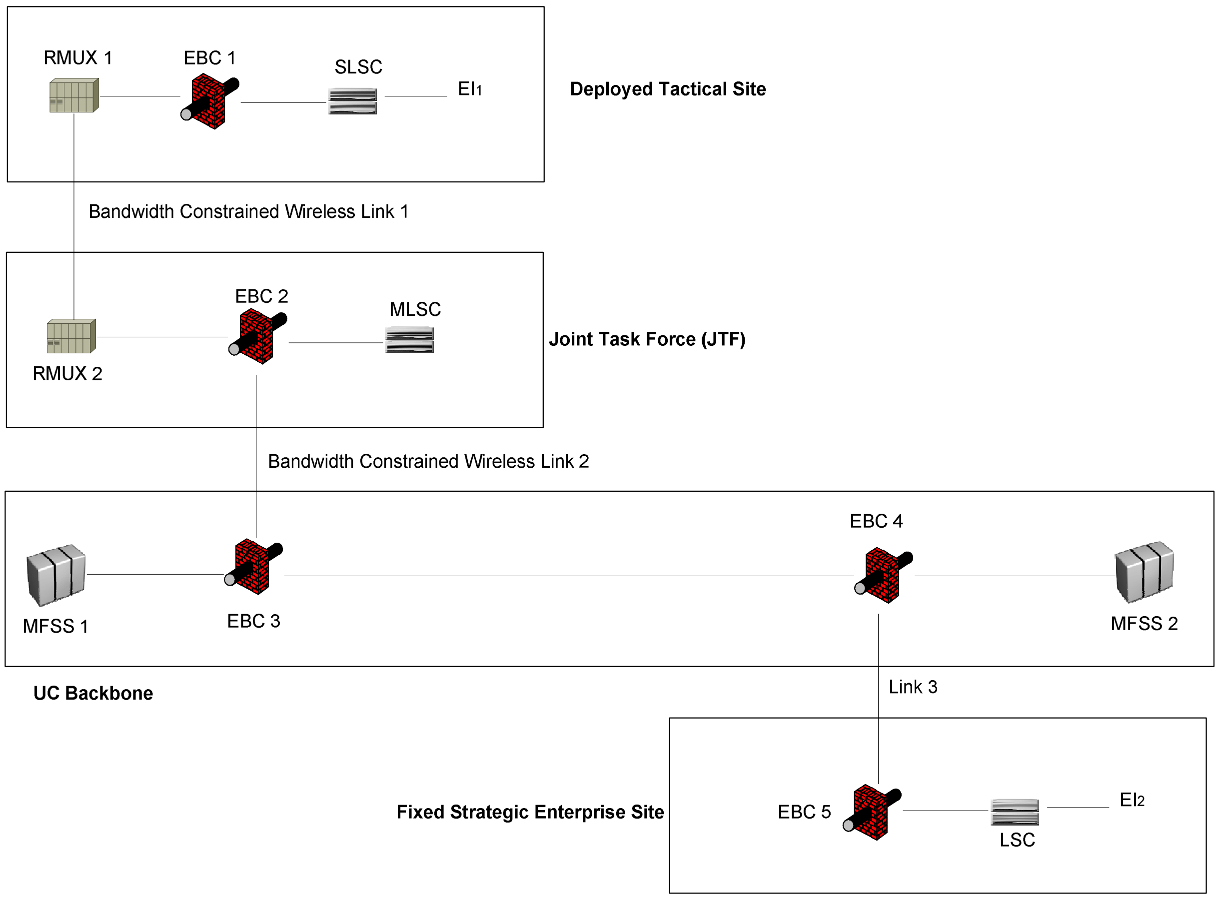


Figure 6.1.8-9 Deployed Site to Enterprise Site via JTF and UC Backbone Topology Example

Upon receipt of the AS SIP INVITE message Subtended LCS 1 and the Master LSC both perform DASAC processing for UC Assured Service calls on bandwidth constrained Link 1 shown in Figure 6.1.8-9, Deployed Site to Enterprise Site via JTF and UC Backbone Topology Example. The Master LSC and the MFSS 1 both perform DASAC processing for UC Assured Service calls on bandwidth constrained Link 2. The LSC with policing from MFSS 2 performs traditional ASAC processing on Link 3. AS SIP message flow for this case in shown in Figure 6.1.8-10.

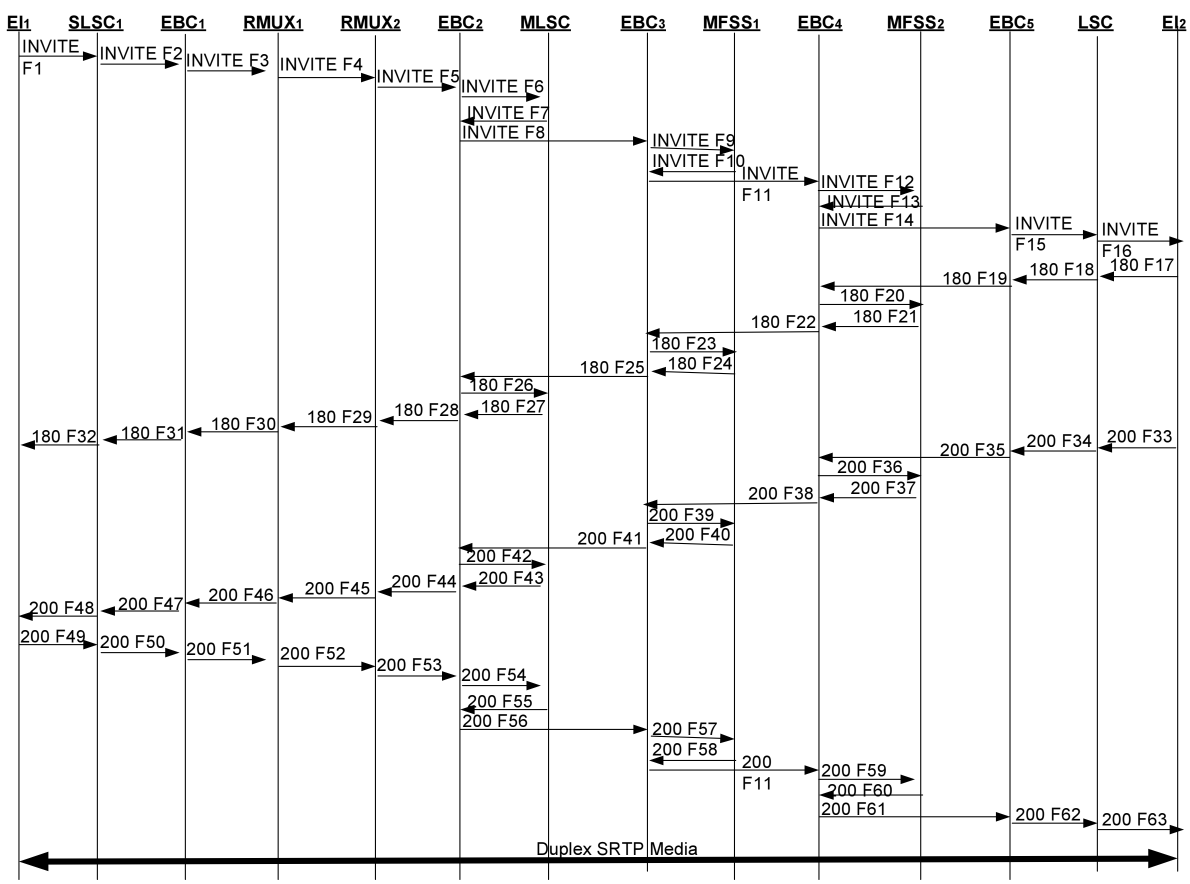


Figure 6.1.8-10 Basic Session Setup Deployed Site to Enterprise Site via JTF

6.1.8.2.5 Deployed Site to Enterprise Via JTF SRTP Flows

The SRTP flows are from end instrument to end instrument but they transit EBCs 1, 2, 3, 4 and 5 as well as the Real-time Muxes 1 and 2 as shown in Figure 6.1.8-11.

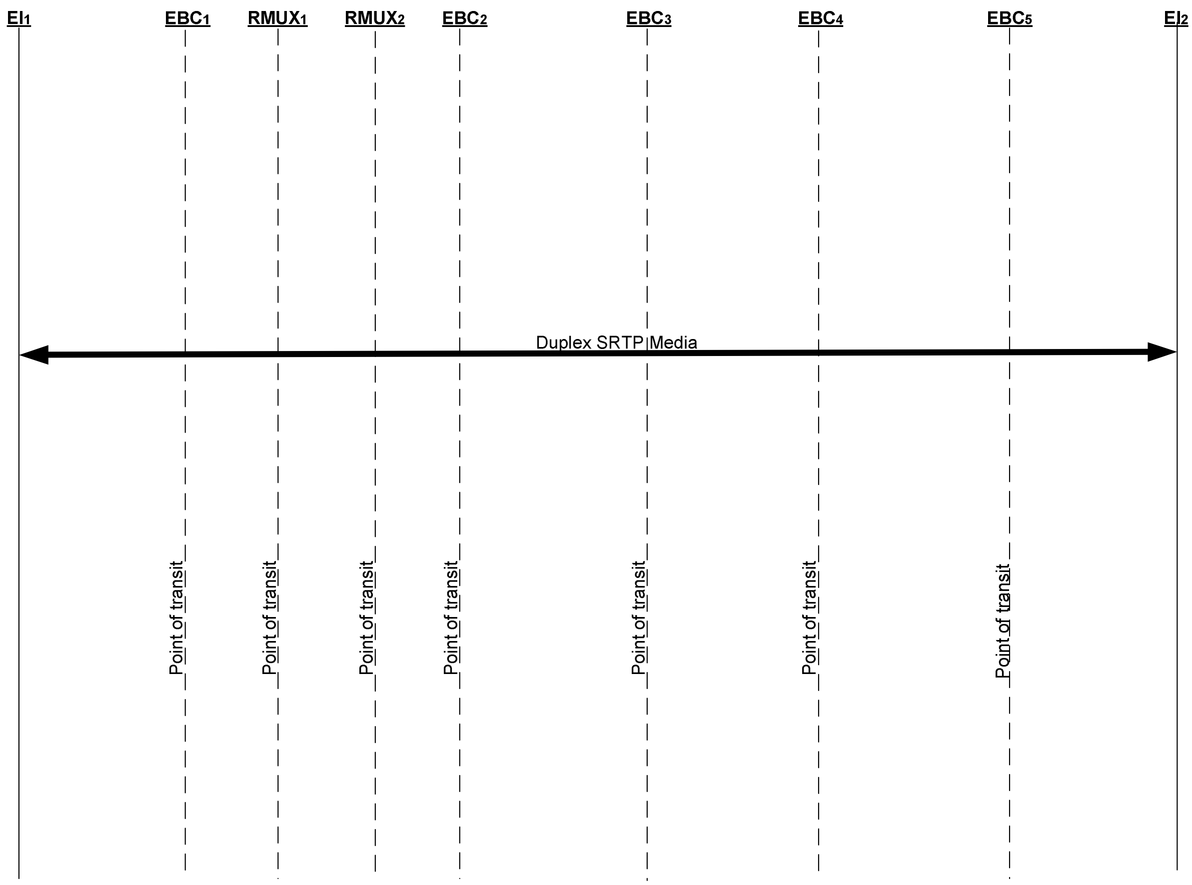


Figure 6.1.8-11 Deployed Site to Enterprise Site via JTF SRTP Flows

6.1.8.2.6 Deployed Site to Enterprise Via JTF Session Teardown

Upon receipt of the AS SIP 200 OK message, the LSC and the MFFS 2 both perform ASAC processing for UC Assured Service calls on Link 3 (see Figure 6.1.8-9). The Master LSC with MFSS 1 policing performs DASAC processing for Link 2. The Master LSC and the Subtended LSC perform DASAC processing for bandwidth constrained Link 1. AS SIP message flow is shown in Figure 6.1.8-12.

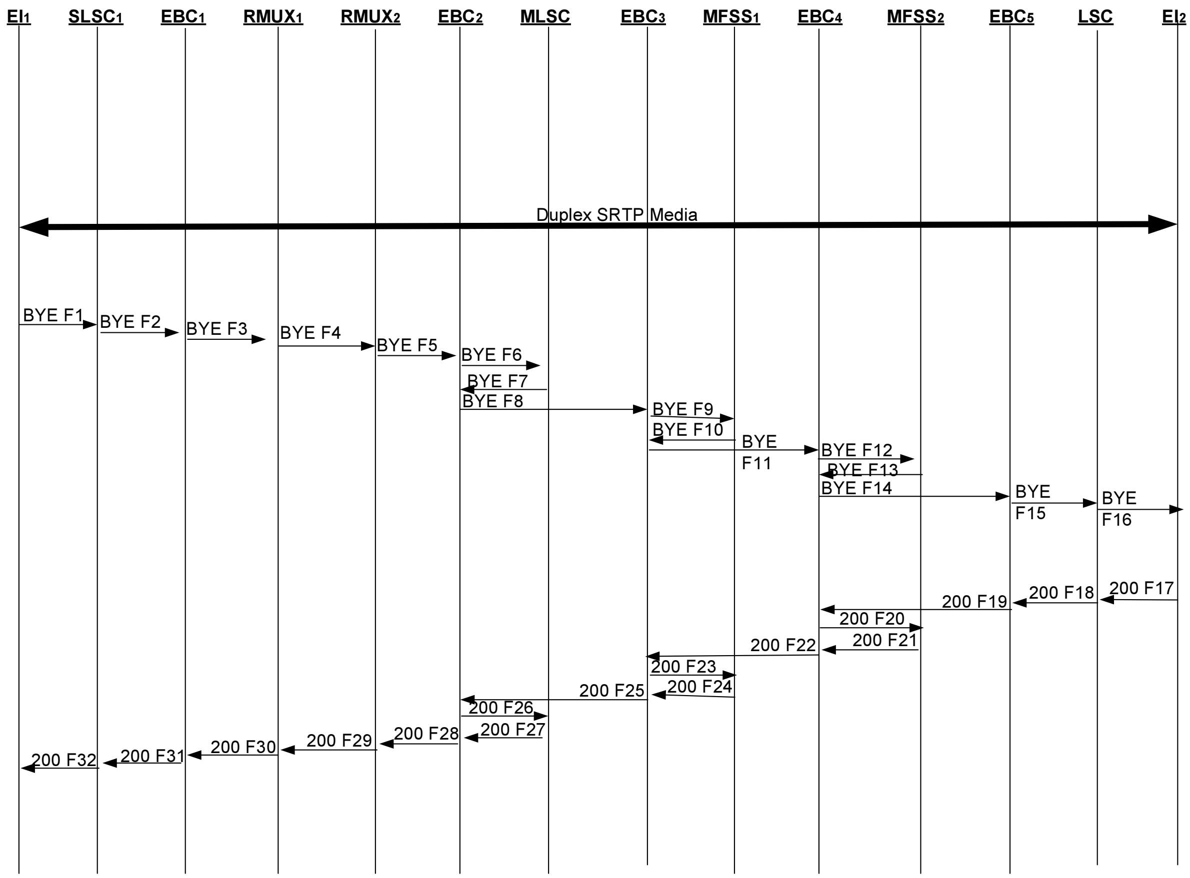


Figure 6.1.8-12 Deployed Site to Enterprise Site via JTF Session Teardown

6.1.8.3 Deployed Diagnostics

Since deployed calls will typically span multiple deployed links, there are benefits to communicating and recording which link was responsible for a given precedence-based block or preemption. These benefits include: fine tuning DASAC budget management, troubleshooting, network load balancing, alternate AS SIP routing, etc. The combination of the Master LSC's and Subtended LSC's CCA IDs or the combination of the Master LSC and the MFSS or WAN SS CCA IDs can be used to identify a link.

[Required Deployed LSC: Communicating Causal Link for Blocks and Preemptions]
The "UC Reason Header for Network Preemption" specified in section 5.3.4.10.3.1.4a shall -be extended to read as follows:

> Reason: preemption; cause=5; text="Network Preemption" SP CCA ID SP CCA ID
>
> where the two CCA IDs represent the two elements (Master LSC, Subtended LSC, MFSS, or WAN SS) that administer the DASAC budget the link.

This Reason Header shall be populated in the 488 (Not Accepted Here) response for a DASAC based block and in the BYE request for a DASAC based preemption. This Reason Header will also be populated in a CANCEL request for an abandoned session setup attempt due to precedence based DASAC block for a session in the middle of session establishment. For these events the Reason Header will also be recorded in the LSCs' CDRs.

The network must continuously monitor its health. Tactical LSCs must periodically confirm the status of active calls.

[Required Deployed LSC: Deployed LSCs Monitoring Deployed LSCs] The Master LSC shall send an OPTIONS request to its subtended LSCs on a configurable basis (default 45 seconds, minimal time interval 35 seconds). The Subtended LSC shall send an OPTIONS request to its Master LSC on a configurable basis (default 45 seconds, minimal time interval 35 seconds). If the LSC does not receive a 200 OK response to its OPTIONS request before the configurable number of successive requests (default equals 2), then the non responding LSC shall be identified in an alarm and in the system log. Once alarmed, the nonresponsive LSC shall not be queried again with the OPTIONS request until reconfigured to do so.

[Required Deployed LSC: Deployed LSCs Monitoring Active Calls] When the DASAC budget processing indicates that a link is supporting one or more calls, the LSC will send an OPTIONS request to all engaged end instrument on a configurable basis (default 5 minutes, minimal time interval 3 minutes). If the LSC does not receive a 200 OK response to its OPTIONS request before the configurable number of successive requests (default equals 2), the call is torn down

6.1.8.4 Dynamic ASAC Requirements for Deployed LSCs

6.1.8.4.1 Dynamic ASAC for Deployed LSCs

This section defines requirements for providing dynamic assured services admission control (DASAC) capability for LSCs that operate within a deployed LAN. DASAC enables an LSC to admit, block or preempt new voice and video calls based on the communications capacity (bits/sec) required for the call and the link capacity available to support the call. DASAC will augment the current ASAC approach in which LSCs admit calls based on a call budget. DASAC will be applied independently to voice and video calls.

6.1.3.4.2 DASAC for Voice

The method for ASAC described in UCR 2008 Change 1 is based on a call budget for each communications link supported by an LSC. The call budget assumes that each call utilizes the same amount of link capacity. The call budget per link is calculated by dividing the link capacity by the data rate required per call and then rounding down to the nearest integer. For example if the link is 512 Kbps and each call is assumed to require 110 Kbps, the call budget will be 4. The LSC tracks the number of calls in progress on a link and only admits a new call if the number of calls in progress is less than the call budget for the link, or there is a lower precedence call that can be preempted to support the new, higher precedence call ~~(see Section ---- regarding precedence and preemption)~~.

In current practice all ~~UC~~VoIP calls are assumed to require 110 Kbps for admission. This number is based on the assumption that the call will employ a G.711 codec, will be encapsulated in an IP packet and might also be encapsulated in an IP tunnel. These are reasonable assumptions in a strategic enterprise environment, but are too conservative to use in a deployed environment, where lower bit rate codecs are used and link capacity is limited. Deployed end instruments (EIs) employ a wide array of codecs - some of which operate as low as 2.4 kbps - to maximize the number of calls on a low capacity link. Therefore the bit-rate required for a call is likely to differ between deployed EIs and possibly within an EI on a call by call basis. The difference depends upon codec negotiation with the other EI(s) participating in the call.

Parameter determination for DASAC can be quite complex. Some call packets might be tunneled over the link, others might not be, others might have header compression and some call packets might be aggregated in a voice multiplexer ~~(see Section -----) also called a "voice mux"~~. Engineering analysis and traffic analysis are required to determine the overheads on the link. The deployed LSC must analyze each call to determine which overheads are appropriate, and the codec rate and packets per second negotiated between the EIs involved in the call. This rate could change during a call, a factor which must be monitored by the LSC.

To facilitate implementation of the DASAC capability, two modes shall be defined:

- The Basic DASAC Mode, which does not take into consideration the impact of header compression or VM even if these features exist on the link.
- The Advanced DASAC Mode, which does take these features into consideration.

6.1.3.4.2.1 Basic Voice DASAC

Required: Deployed LSCs shall manage DASAC to a budget in a manner similar to that described in Section 4.~~5~~4.1.1.5 (ASAC Component) with the following modification: All LSCs that support deployed telephone end instruments shall manage the DASAC budget based on the required capacity for a call (as measured in bits per second) and the available capacity on the target link supporting that call.

Required: DASAC shall employ a method of establishing and managing the budget per target link (to be called the" deployed call budget") based on the following parameters:

1. End Instrument Call Capacity (EICC) - the link capacity in bits per second required for the call. EICC shall be computed by the LSC each time it detects signaling for a new call or a change in codec parameters for an ongoing call.

2. Transmission Link Call Capacity (TLCC) - the capacity (bits per second) of the target link allocated for UC telephone calls. TLCC is a pre-provisioned parameter entered for each target link via network management commands. The TLCC does not include an allocation for call signaling. Call signaling must be provisioned separately as part of traffic engineering for the target link.
3. Available Link Call Capacity (AVCC) - the capacity (bits per second) currently available for calls on the target link. AVCC shall be calculated each time during:
   - The session establishment AS SIP dialog (specifically the AS SIP message containing the SDP answer)
   - Mid-session re-INVITE dialog based on a mid-call codec change (specifically the AS SIP message containing the new SDP answer to the new offer)
   - Call tear down (specifically based on LSC detecting the AS SIP 200 OK for the BYE)

AVCC is calculated as follows:

AVCC = TLCC – Sum of EICCs for all calls in progress and in the process of being established.

Determination of TLCC depends on:

1. The allocation of capacity to the router queue supporting the voice traffic under the control of the LSC (typically the EF queue within a customer edge router)
2. The portion of that capacity which is reserved for voice applications that are not under control of the LSC[1]

Figure 6.1.8-13 AS SIP Triggers for AVCC below illustrates the AS SIP triggers for the AVCC calculations. For reasons of simplification it assumes the end instruments are AS SIP enabled.

[1] It is possible that there will be non-UC applications supported in the same router queues that support DASAC flows. These will not be under the control of the LSC. Traffic engineering must account for the capacity that is guaranteed to these flows. This value must be subtracted from the total capacity allocated to the router queue. The non-UC traffic must be controlled via admission control or router policing to ensure that the capacity allocated to the UC traffic is protected.

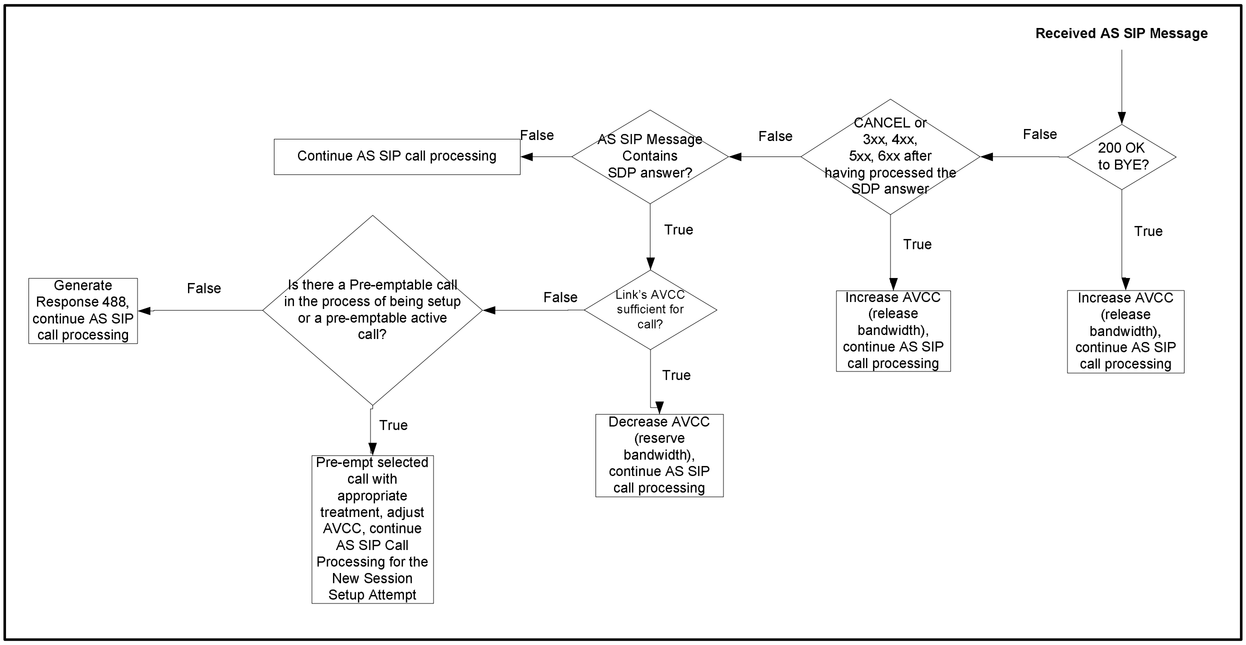


Figure 6.1.8-13 AS SIP Triggers for AVCC

When a "200 OK to a BYE" is received by the deployed LSC, the bandwidth previously reserved for this call is released and thereby the AVCC is increased.

The "SDP Answer" message indicates the results of the codec negotiation between the end instruments involved in the call request. The LSC processes the SDP Answer to determine if there is sufficient capacity to support the new call. If so, the LSC will reserve bandwidth for the call and continue with AS SIP processing. If a Cancel or a 3xx, 4xx, 5xx, or 6xx message is received after the SDP Answer is processed but before the call setup is completed, the reserved capacity will be released and the AVCC increased accordingly.

-If after receiving a SDP Answer, the LSC determines that there is not sufficient bandwidth for the new call, the LSC will review active calls, and calls with previously reserved bandwidth that are still in the set up process, to determine if any of these calls are eligible for pre-emption. To be pre-empted such calls must have a lower precedence than the new call and must release. sufficient bandwidth to support the new call. If both on-going and in process calls are eligible for preemption, the LSC shall preempt one of the in process calls.

If no preemptible calls exist a "488 Response" is generated, which will lead to blockage of the new call. Note that if pre-emption occurs and the new call is setup, the resulting AVCC may be larger or smaller than before the pre-emption. If the pre-empting call's bandwidth requirements are less than that of the pre-empted call, the AVCC increases. If the pre-empting call's bandwidth requirements are more than that of the pre-empted call, the AVCC decreases.

[Required]: The LSC shall support separate TLCC and AVCC parameters based on direction of a call ~~(see Section ----)~~. A call request must have sufficient capacity in both directions, either directly or via preemption, in order to be accepted as a new call

[Required]: The LSC shall be able to support TLCC and AVCC for multiple links.

[Required]: The LSC shall be able to support TLCC and AVCC for three-way calls.

[Required]: The LSC shall be provisioned with call budget parameters which provide an absolute limit on the number of calls accepted in either direction for each link ~~(see Section ----)~~.

The EICC for an individual end instrument could vary from call to call based on the Codec Rate and Packets per Second rate negotiated during the Session Description Protocol (SDP) message exchange that takes place during call set up ~~(see Section ----)~~.

[Required]: Calculation of EICC shall be based on the factors shown in Table 6.1.8-1. Parameters 1-3 are dynamic; the LSC must calculate these parameters on a call by call basis. Parameters 4-7 are pre-loaded into the LSC based on traffic engineering analysis of the link.

[Required]: The LSC shall, on a call-by-call basis, scan the SDP messages, extract EICC Codec Rate and Packets per Second and store these parameters for each call in progress within a DASAC data base. The LSC shall be able to support all Codecs defined in Section ~~-----------~~ 5.3.2.6 End Instruments.

[Required] If the LSC cannot determine either Codec Rate or Packets per Second, the default EICC for a call shall be 110 Kbps. In such case the LSC shall create an error message in the call detail record and trigger a network management alarm indicating that a fault condition has occurred.

[Required] If the values of parameters 4 through 7 are not explicitly entered in the data base, the LSC shall use the default parameters listed in Table 6.1.8-1.

[Required]: The LSC shall not use silence suppression, also known as voice activity detection, as a factor in calculating EICC.

Examples of EICC calculation are provided in Tables 6.1.8-2 and 6.1.8-3. The environment for these examples is shown in Figure 6.1.8-14. The environment consists of 20 V~~O~~oIP phones, each of which can support G711, G729, G723, and 2400 bps MELPe codecs. The TLCC for the controlled link is 256 Kbps full duplex. There is no tunnel in Example 1; there is a HAIPE Tunnel in Example 2.

The first seven parameters in Tables 6.1.8-2 and 6.1.8-3 are the same as those in Table 6.1.8-1. Notional values are provided in Tables 6.1.8-2 and 6.1.8-3. The other line items represent overheads and traffic flows calculated from the first 7 parameters. Tables 6.1.8-2 and 6.1.8-3 also show the TLCC and AVCC.

Table 6.1.8-1 Per Link EICC Estimation Parameters per Codec Class

Codec Class is determined by Codec Rate and Packet Rate; LSC shall maintain the following parameters per Codec Class per monitored link.			
#	Parameter	Source	Comment
1	**Codec Rate** in bits per second (bps)	LSC extracts from SDP message - stored per Codec Class	Could change on a call by call basis per End Instrument
2	**Packet Rate** in packets per second (pps)	LSC extracts from information in SDP message	Could change on a call by call basis per End Instrument
3	**Number of Voice Sessions in Progress**	Number of calls in progress for this Codec Class Running account kept by LSC	**Initial value = 0** Incremented upon successful call connection Decremented upon successful call completion
4	**Tunnel Overhead Factor** in bytes	Pre-provisioned and entered into LSC	Indicates the number of overhead bytes that must be added to the IP packet size to account for encryption or other types of tunnels. If some calls are tunneled and others not use the number of bytes associated with largest overhead tunnel. **Default is 100 bytes.**
5	**IP Overhead** in bytes	Pre-provisioned and entered into LSC ; Includes IP, UDP, RTP overhead associated with packet flow over the target link	If any calls are IPv6 use 60 bytes If all calls are IPv4 use 40 bytes **Default is 60 bytes.**
6	**Layer 2 Overhead** in bytes	Pre-provisioned and entered into	Sized according to layer protocol used on link – this parameter is the same for all packets in all Codec Classes. **Default is 20 bytes.**
7	**Safety Factor** in Per Cent	Pre-provisioned and entered into LSC.	This parameter is used to provide a margin of error for the EICC calculation. **Default is 10%.**

Example 1 shows the AVCC at a point in time where there are 8 calls in progress. Five of these are MELPe calls[2] with one call each for the other codecs. The total EICC for these calls is 176.8 Kbps, as shown in Table 6.1.8-2. The AVCC is 79.2 Kbps, based on a TLCC of 256 Kbps. In this case the LSC could admit a new call using any of the Codec Types except G711. If the next call offers a G711 codec, the LSC must block the call unless a there is a lower precedence call that can be preempted.

Example 2 shows the AVCC calculation for the case where a HAIPE is used to encrypt packets on the target link. In this example there are 7 calls in progress: 5 MELP calls, one call for G723.1 and one call for G729. The AVCC calculation takes place just after an INVITE for a new G711 call is generated. The LSC calculates the AVCC for the G711 call at negative 7.3

[2] In these examples the calculations for MELPe capacity requirements include an overhead factor of 1.037 to account for padding the MELP codec bits to fit into a 7 octet voice sample.

Kbps (see Table 6.1.8-3). The LSC will reject the new call if it cannot preempt one of the existing calls.

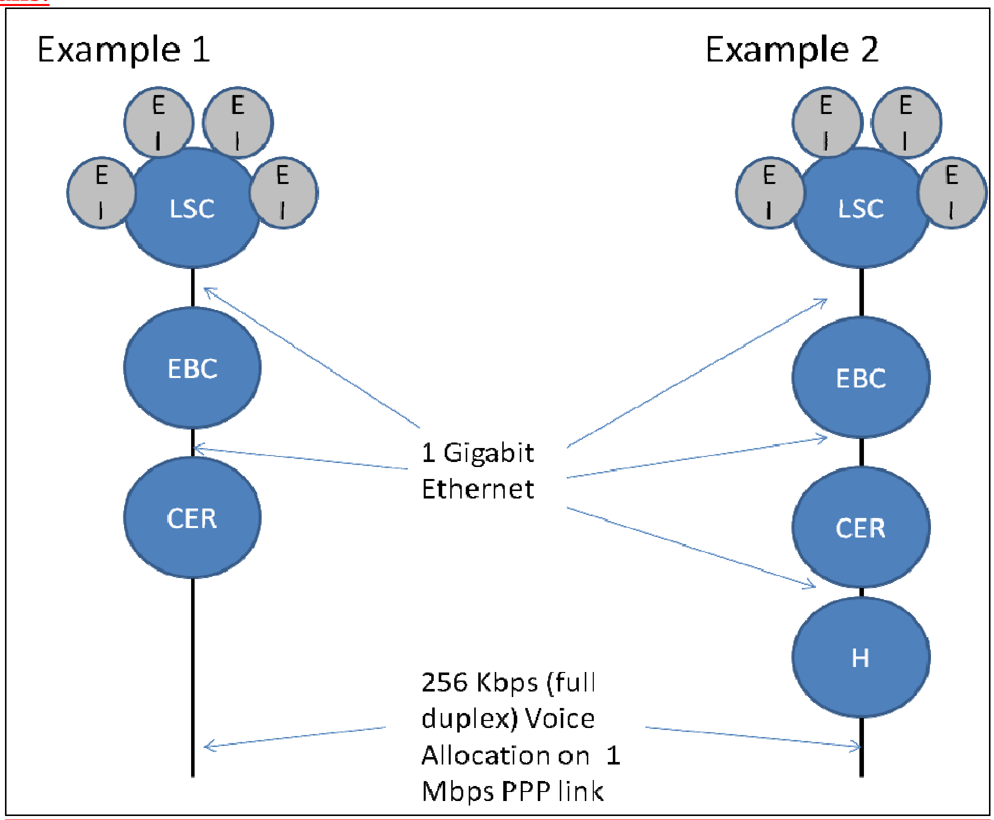


Figure 6.1.8-14   Notional System Architecture for Examples 1 and 2

Table 6.1.8-2 Example 1: Current Call Status (No HAIPE Case)

ID				MELPe	G723.1	G729	G711
	TLCC	IPV4					
		TLCC=	256	Kbps			
ID	Codec Type			MELPe	G723.1	G729	G711
1	Codec Rate	Kbps		2.4	5.3	8	64
2	Packet Rate	Packets per Second		11.1	33.3	50.0	50.0
3	Number of Voice Sessions in Progress			5	1	1	1
4	Tunnel Overhead	Bytes		0	0	0	0
5	IP Overhead	Bytes		40	40	40	40
6	Layer 2 Overhead Rate	Bytes		7	7	7	7
7	Safety Factor	%		10%	10%	10%	10%
8	Payload Size	Bytes		28	20	20	160
9	Packet Size	Bytes		68	60	60	200
10	Packet Rate	Kbps		6.0	16.0	24.0	80.0
11	Layer 2 Overhead	Kbps		0.6	1.9	2.8	2.8
12	Average Data Rate for Payload and Overhead	Kbps		6.7	17.8	26.8	82.8
13	EICC (including Safety Factor) per call	Kbps		7.3	19.6	29.5	91.1
14	Total EICC for all calls in Codec Group	Kbps		36.7	19.6	29.5	91.1
15	Total EICC for all calls on link	Kbps		176.8			
	Grand Total Calls			8			
	AVCC	Kbps		79.2			

Table 6.1.8-3 Example 2: AVVC Calculation Assuming the G711 Call is New (HAIPE Case)

		HAIPE TUNNEL					
		IPV4					
		TLCC=	256	Kbps			
ID	Codec Type			MELPe	G723	G729	G711
1	Codec Rate	Kbps		2.4	5.3	8	64
2	Packet Rate	Packets per Second		11.1	33.3	50.0	50.0
3	Number of Voice Sessions in Progress			5	1	1	1
4	Tunnel Overhead	Bytes		52	52	52	52
5	IP Overhead	Bytes		40	40	40	40
6	Layer 2 Overhead	Bytes		7	7	7	7
7	Safety Factor	%		10%	10%	10%	10%
8	Payload Size	Bytes		28	20	20	160
9	Packet Size	Bytes		120	112	112	252
10	Packet Rate	Kbps		10.7	29.8	44.8	100.8
11	Layer 2 Overhead Rate	Kbps		0.6	1.9	2.8	2.8
12	Average Data Rate for Payload and Overhead	Kbps		11.3	31.7	47.6	103.6
13	EICC (including Safety Factor) per call	Kbps		12.4	34.9	52.4	114.0
14	Total EICC for all calls in Codec Group	Kbps		62.1	34.9	52.4	114.0
15	Total EICC for all calls on link	Kbps		263.3			
	Grand Total Calls			8			
	AVCC	Kbps		-7.3			

6.1.3.4.2.2 Advanced DASAC

Advanced DASAC incorporates additional parameters and calculations to account for VM and header compression.

[Conditional]: The LSC shall support DASAC for links where all assured voice traffic is processed by a voice multiplexor prior to transmission on the link. The LSC shall use the parameters in Table 6.1.8-1, augmented by the parameters in Table 4, to calculate the total EICC and AVCC. The parameter values in Table 6.1.8-4 shall be determined prior to operation and loaded into the LSC data base. The values shall be chosen conservatively to that there is no case where more calls can admitted than can be supported by the target link.

Table 6.1.8-4 VM Parameters

#	Parameter	Source	Comment
1	**Voice Mux Overhead per Packet** (bytes)	Pre-provisioned and entered into LSC	This parameter is used on a per packet basis
2	**Overhead per Voice Mux Sample** (Bytes)	Pre-provisioned and entered into LSC	This parameter is an overhead that is applied to each voice sample bundled in a output voice packet
3	**Payload Size (Bytes)**	Calculated by LSC based on Codec Rate and Packet Rate	

Example 3, in Table 6.1.8-5, shows the case where a voice multiplexer is used to reduce the EICC per call. The environment for this example is shown in Figure 6.1.8-15. The TLCC is 256 Kbps (full duplex). The network contains a HAIPE device.

Table 6.1.8-5 Example 3 – Use of Voice Mux With a HAIPE Tunnel

	Voice Mux Calls	IPV4					
	Per Codec Type Calculations	TLCC=	**256**	Kbps			
ID	Codec Type		Packet	MELPe	G723	G729	G711
1	Codec Rate	Kbps		***2.4***	***6.4***	***8***	***64***
2	Packet Rate	PPS		***11.1***	***33.3***	***50.0***	***50.0***
3	Number of Voice Sessions in Progress			***5***	***1***	***1***	***1***
4	Overhead for voice mux sample	Bytes		**7**	7	7	7
5	Payload size	Bytes		28	24	20	160
6	Payload traffic rate	Kbps		2.49	6.40	8.00	64.00
7	Voice mux overhead traffic rate	Kbps		3.1	1.9	2.8	2.8
8	Voice mux and payload traffic rate	Kbps	91.5	5.6	8.3	10.8	66.8
	Per Packet Overhead Calculation						
9	Tunnel overhead	Bytes	**52**				
10	IP overhead	Bytes	**28**		MELPe Codec rate		
11	Voice mux overhead per packet	Bytes	**4**		adjustment factor =		**1.037**
12	Layer 2 overhead	Bytes	**12**				
13	Safety Factor	%	**10%**		Packet rate is calculated		
14	Total per packet overhead rate	Kbps	42.24		as the maximum Packet		
15	Total EICC for all calls in progress	Kbps	133.7		Rate above (ID=2)		
	Grand Total Calls		8				
	AVCC	Kbps	122.3				

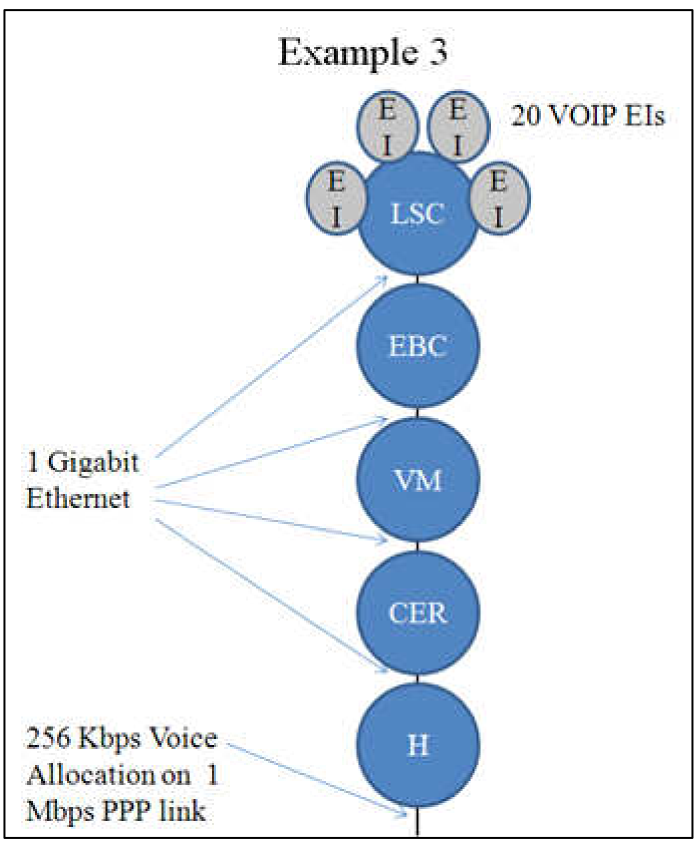


Figure 6.1.8-15 - Notional System Architecture for Example 3 (Voice Mux and HAIPE Tunnel)

Table 6.1.8-5 shows the AVCC at a point in time where there are 5 MELPe calls and one call each based on G723.1, G729, and G711 codecs. The calculation takes into account two types of overhead; one for each packet transmitted by the voice multiplexer; the other for each voice sample in the output packet.

The per-packet overhead consists of the IP, tunnel and voice mux byte overheads for each output packet. This overhead is multiplied by the output packet rate to determine the overhead in Kbps. The output packet rate is set at the highest rate of the codecs supported by the end instruments on the LSC side of the target link. The voice sample overhead is the number of bytes that the voice mux appends to each voice sample included in the output packet. The number of bytes per sample is multiplied by the voice sample rate of the input packets, to determine the overhead in Kbps.

The EICC for this example is 133.7 Kbps, as shown in Table 6.1.8-5. The AVCC is 122.3 Kbps, based on a TLCC of 256 Kbps. In this case the LSC could admit a new call from any of the Codec Types. Compared to Example 2 the voice mux reduces the bandwidth demand from 263.3 Kbps to 133.7 Kbps, based on the notional numbers used in the examples.

[Conditional]: The LSC shall support DASAC for links where all assured voice traffic is processed by a header compression mechanism in the CER. The LSC shall use the Table 6.1.8-1 parameters, with the IP header parameter modified to reflect the smaller, compressed header. The parameter shall be tailored to the specific header compression algorithm used in the CER with sufficient safety factor to account for the header compression signaling and for the occasional occurrence of full headers.

An example calculation for header compression is shown in Table 6.1.8-6. Example 4 is based on the approach used in Example 2. The IP overhead parameter has been modified to account for a header compression mechanism that, on average, transmits 95% of packets with a compressed header of two bytes, and 5% of packets with a full IP, UDP, and RTP header of 40 bytes. This gives an average header size of 3.9 bytes, which has been rounded up to 5 bytes to provide a margin of safety.

The AVCC is 50.9 Kbps which would enable the LSC to accept any new call request without preemption, except for a call that requires a G711 codec.

ID	Codec Type			MELPe	G723	G729	G711
		HAIPE TUNNEL					
		IPV4					
		TLCC=	**256**	Kbps			
1	Codec Rate	Kbps		***2.4***	***5.3***	***8***	***64***
2	Packet Rate	Packets per Second		***11.1***	***33.3***	***50.0***	***50.0***
3	Number of Voice Sessions in Progress			***5***	***1***	***1***	***1***
4	Tunnel Overhead	Bytes		***52***	***52***	***52***	***52***
5	IP Overhead	Bytes		**5**	5	5	5
6	Layer 2 Overhead	Bytes		**7**	**7**	**7**	**7**
7	Safety Factor	%		**10%**	**10%**	**10%**	**10%**
8	Payload Size	Bytes		28	20	20	160
9	Packet Size	Bytes		85	77	77	217
10	Packet Rate	Kbps		7.6	20.5	30.8	86.8
11	Layer 2 Overhead Rate	Kbps		0.6	1.9	2.8	2.8
12	Average Data Rate for Payload and Overhead	Kbps		8.2	22.4	33.6	89.6
13	EICC (including Safety Factor) per call	Kbps		9.0	24.6	37.0	98.6
14	Total EICC for all calls in Codec Group	Kbps		45.0	24.6	37.0	98.6
15	Total EICC for all calls on link	Kbps		205.1			
	Grand Total Calls			8		MELP overhead	
	AVCC	Kbps		50.9		factor =	**1.037**

Table 6.1.8-6 Example 4 – Use of Header Compression With a HAIPE Tunnel

6.1.3.4.3 DASAC for Video

[Required] The LSC shall support DASAC for video calls that use AS SIP signaling. The LSC shall set up a DASAC video budget based on the Kbps allocated across the target link. The LSC shall use the parameters shown in Table 1 to compute the EICC for each new call request to determine the AVCC. The LSC shall admit the call if the AVCC is positive. If the AVCC is negative the LSC shall accept the call only if there is a lower precedence video call that can be preempted to provide the required capacity. If not, the LSC will block the call.

[Conditional]: The DSAC calculation can be modified to take into account the bursty nature of video traffic to facilitate statistical multiplexing of UC video streams. The method for doing so is left to the vendor. This method must ensure that the resulting maximum traffic load does not create an average packet loss greater than one packet per hundred.